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By Marcel van de Gevel

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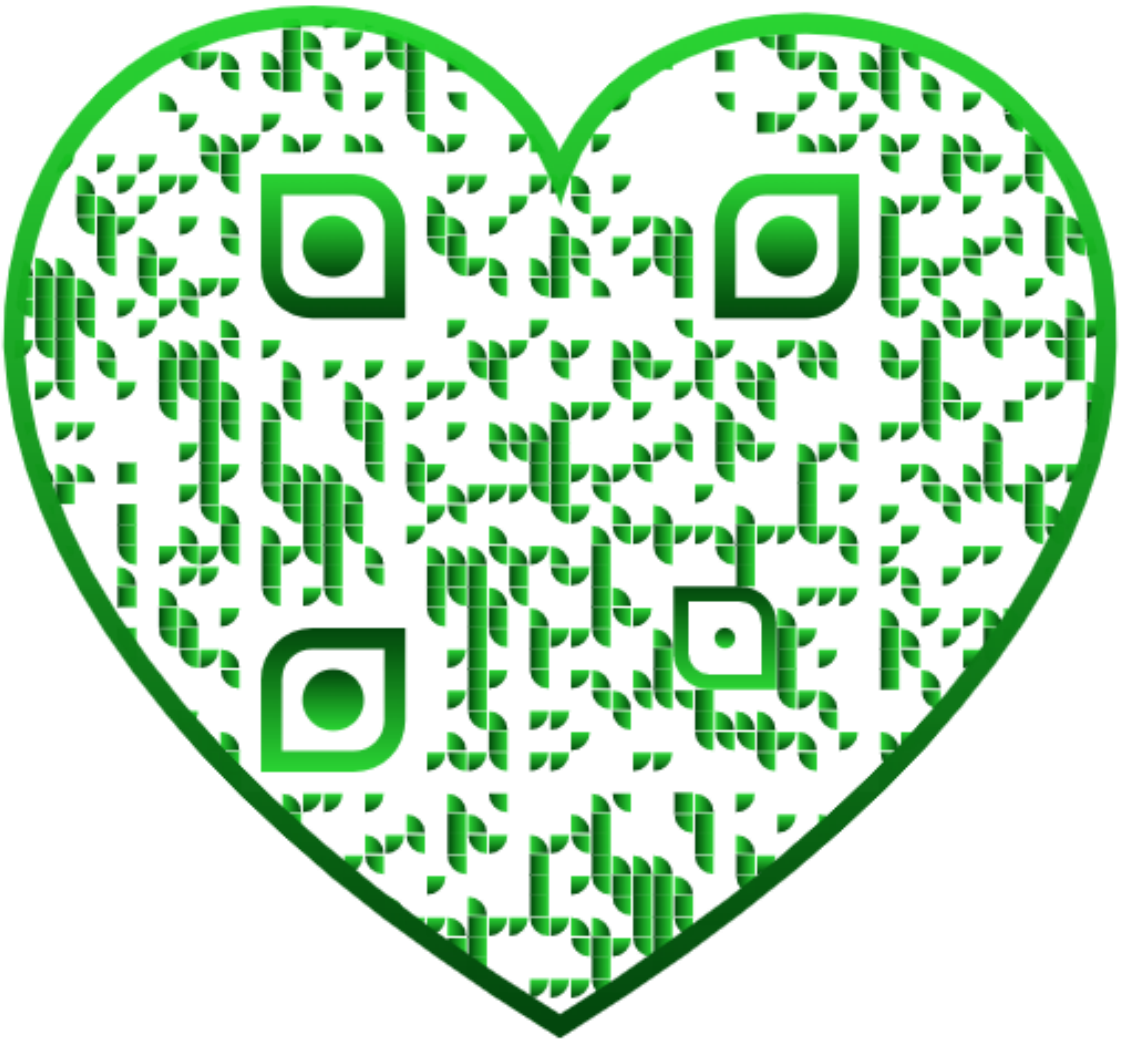

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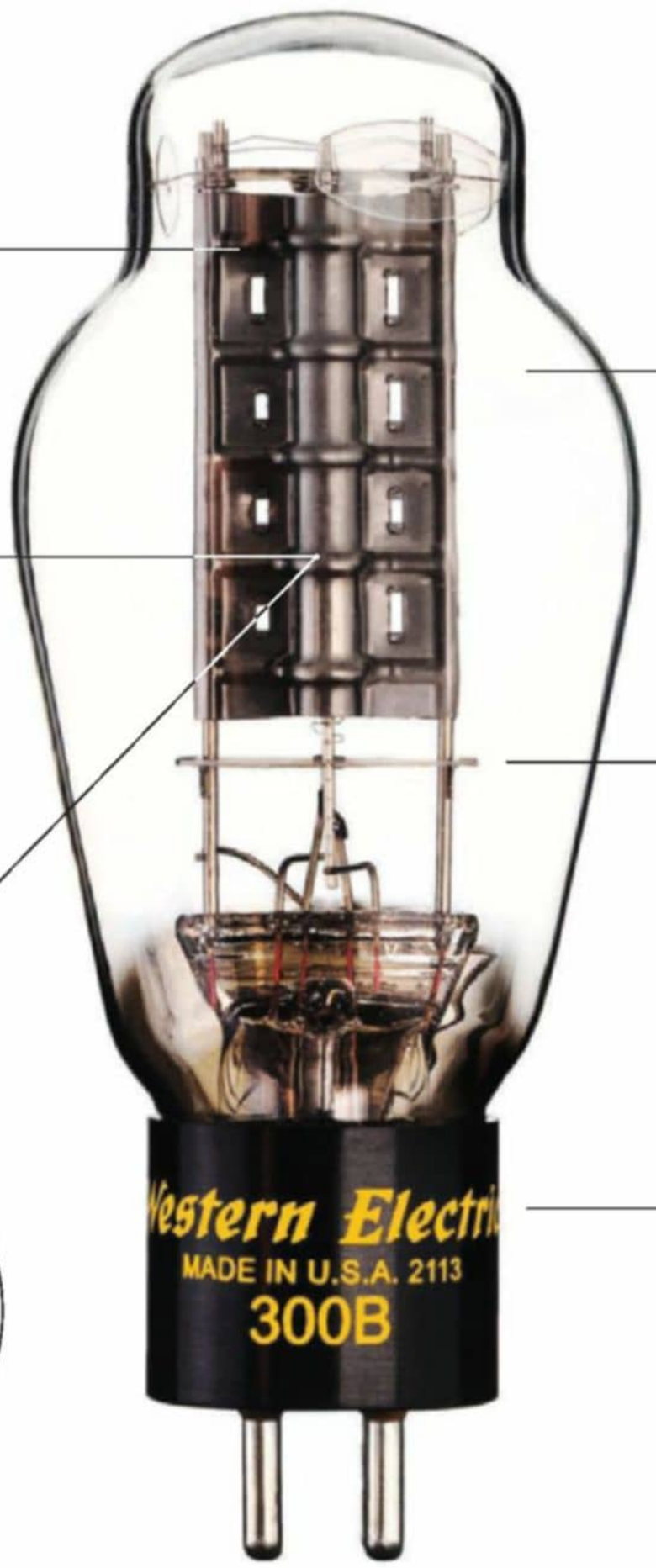

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Invitation-Only Audio History

Since returning from CES 2023, I feel there are some experiences and impressions that are worth leaving here for posterity. After nearly three years with no major trade shows contributing much in terms of surprises and excitement, the annual Las Vegas, NV, event was clearly a great way to kick start the new year. It wasn't a big event for new products launches, nor did it signal the introduction of new technologies of which we weren't aware.

Plenty of new products were already launched in 2022, simply because many companies were forced to postpone their planed introductions for nearly two years due to the global pandemic. And most of the technologies we got to see at CES 2023 were in fact things we have been reporting for some time.

So, why the excitement? Well, it comes from actually being at a show and meeting the people and the companies behind the products and technologies and discussing their plans for the forthcoming months and years. In that regard, CES was a great moment to confirm what is real, what are mainly concepts, and what's smoke and mirrors—and that's precisely what makes trade shows so great.

CES 2023 also confirmed that global business is about to make a comeback, with major differences regarding managing supply chains and sales channels. As I reported in my articles in *The Audio Voice* newsletter—the most important trend at CES is how business is increasingly conducted in private suites and invitation-only presentations, all over Las Vegas, and outside of the regular visitor's eyes. Brands, manufacturers, product managers, design and engineering houses are meeting with technology, parts, and OEM/ODM services suppliers in audio, as in video, IT, accessories, health and wellness products, and now even in automotive. That business returned to CES 2023 even stronger this year, and B2B actually invaded even the large exhibition halls, where multiple industries that have never been involved with CES now go to do business and to benefit from the global media boost.

This makes CES a major show for all consumer industries, including audio, even though the "demonstrations" are less of finished products, and almost always of working concepts and prototypes. For thousands of startups active in every product segment this creates a vibrant environment to showcase innovation. I was particularly glad to see companies showcasing speakers, audio amplifiers, headphones, hearables, and automotive sound, enjoying success and gaining business. Some of those companies were in small standard booths at the Eureka Park—the startup arena. Some startups I had seen in 2020 were already in private suites with manufacturers lining up for meetings.

This is also a problem for the organizers of CES, the Consumer Technology Association, North America's largest technology trade association, which needs to readjust CES and its layout to accommodate this new B2B-oriented model.

For our readers, curious about what we've seen there, the most exciting innovations are not all necessarily audio. Among those are energy solutions from the Swedish company Exeger, the inventor and maker of Powerfoyle, a solar cell flexible material that converts all forms of light into clean, endless energy. Matched with thin and flexible batteries that can withstand more charging cycles than other Lithium-ion batteries (e.g., NGK EnerCera), we will see more and more headphones and even portable speakers that will rarely need to charge from the grid, and low-power devices, such as remote controls that will not use discardable batteries. And I should mention the interesting USB technology updates, including (finally!) an upcoming update to the USB Audio specification, or the many interesting proposals to improve wireless audio transmission, including some progress on Wi-Fi-based implementations (DTS Play-Fi, and LG WOWCAST), new exciting codecs for high-resolution audio, and interesting demonstrations over Ultra Wideband (UWB).

And totally audio-related, the Audioscenic technology that truly delivers personal 3D sound through speaker arrays, using head-tracking, and DSP with AI capabilities. The experience, which adapts in real-time to the listeners position, creates best in class immersive sound. That, together with all the demos of adaptive room correction, from a single omnidirectional smart speaker to multichannel home theater systems, and the exciting active acoustics demonstrations in the automotive environment, are a great indicator of the many fantastic things to come in audio.

And my final recommendation would be to listen to the latest 1000W reference design from GaN Systems and Axign. I believe that you will feel the ground you're standing starting to move.

And no, none of that was showing at the CES show floor, but they are now part of the show's history.

J. Martins
Editor-in-Chief

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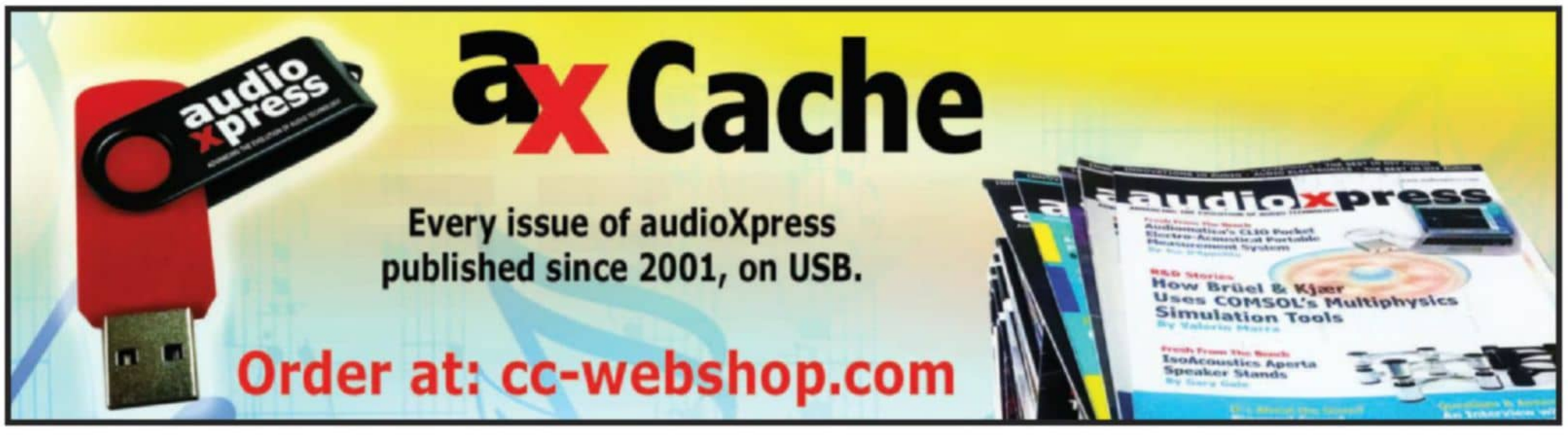
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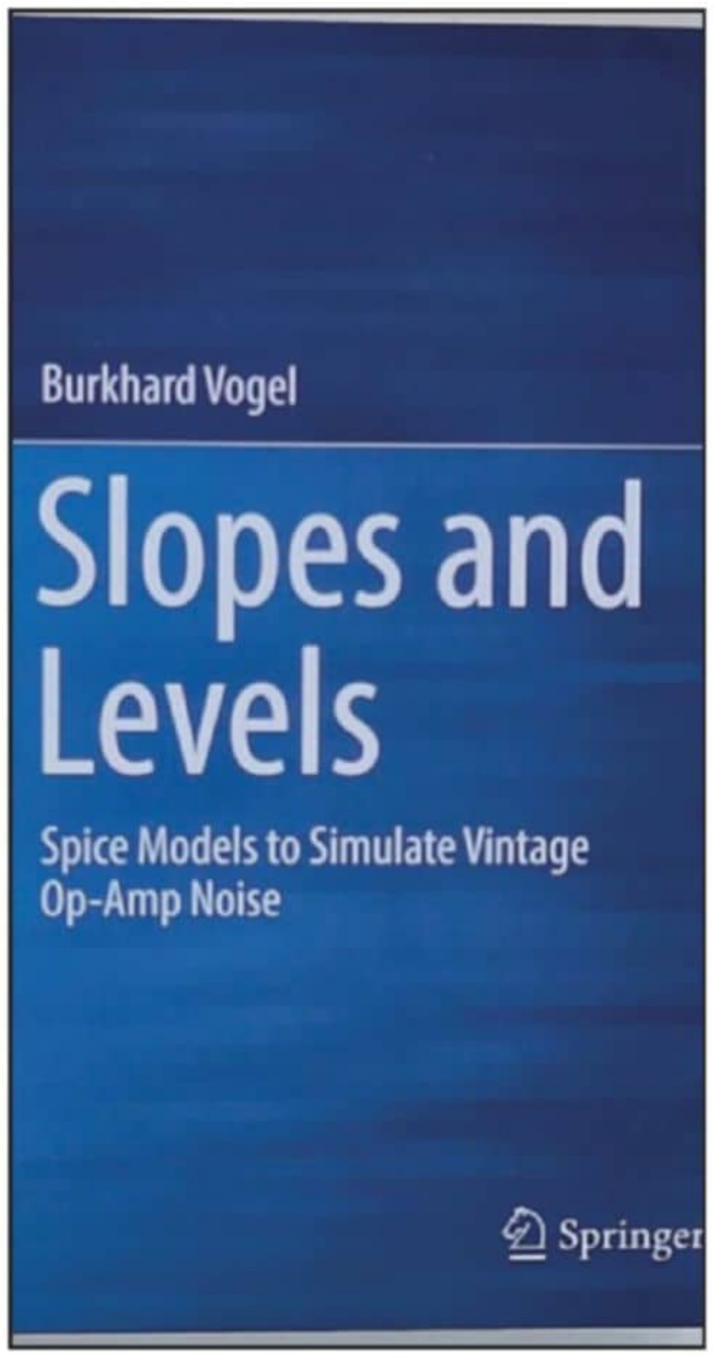

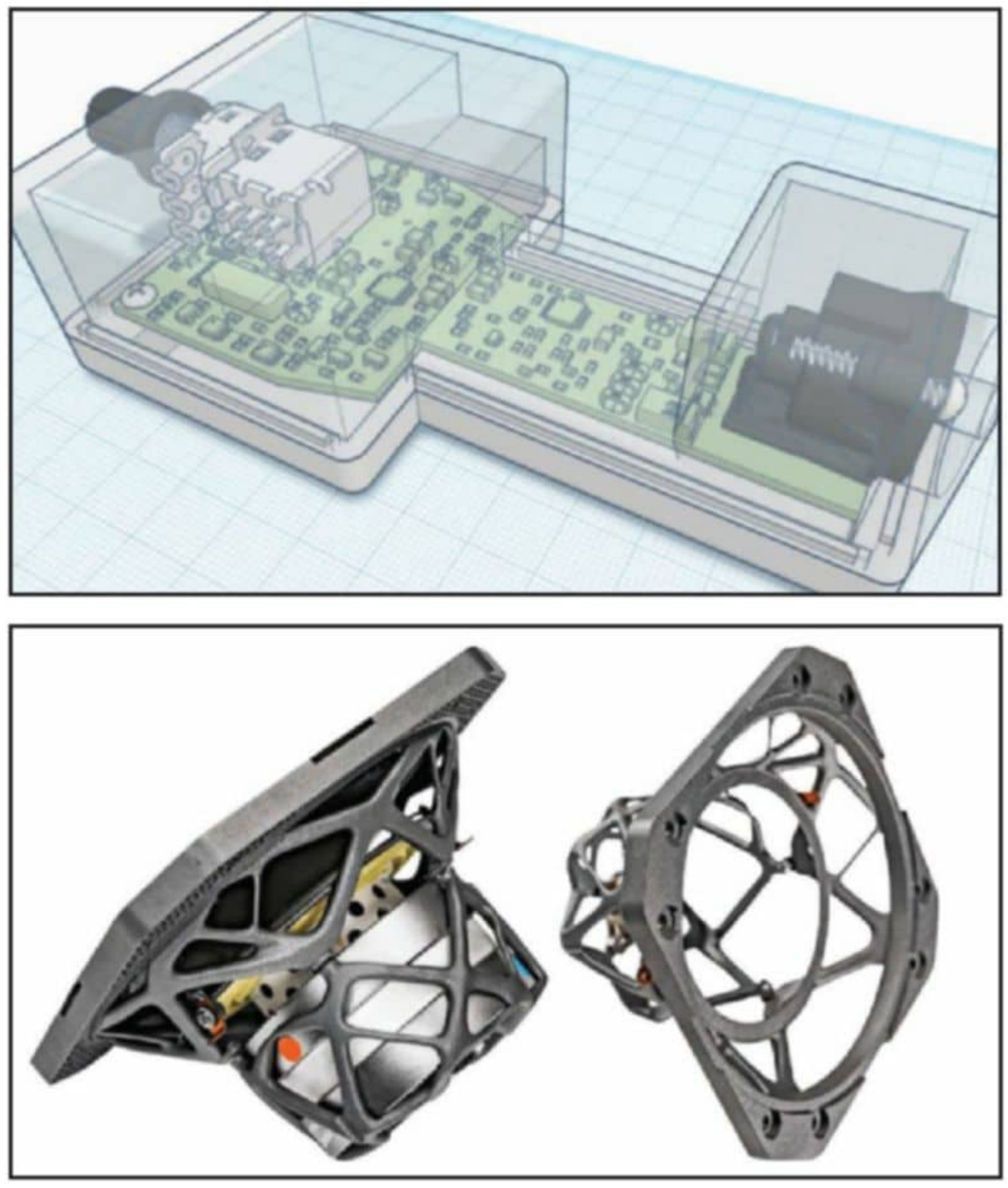

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Fresh From the Bench

Echo AIO

A Cost-Effective Audio Test System

By **Stuart Yaniger**

Headquartered in Santa Barbara, CA, Echo Test + Measurement is a little company with a big impact on audio test and measurement for consumer electronics. Based on its sound card technology, it created a full modular test interface range, which Stuart Yaniger had the opportunity to evaluate.

Audio test and measurement has generally been clearly divided between home and professional users. Professional gear from the likes of Audio Precision or Rohde & Schwarz is rugged, reliable, high performance, and generally easy to set up. The bench instruments are particularly capable of quickly changing over measurements from one type to another and are notable for their versatility, since they may be used for a variety of measurement types and goals, from development to research to product validation.

Production instruments are generally less flexible, but easy to program to allow production staff to perform repetitive tasks such as quality control checks rapidly, consistently, and without having to delve into the engineering. All of this comes at a price, of course, but traceability and the ability to automate are vital for manufacturing.

Small shops and home users have been blessed by the proliferation of high-quality computer sound cards or analog interfaces, depending on whether the target audience is musicians or audio developers, and incredibly versatile measurement software packages. With suitable external electronics (signal conditioners, microphone preamps, power amplifiers, test resistors, and more), a small lab can perform just about any imaginable audio measurement. And do it cheaply—the cost for an excellent sound card plus signal conditioner (the Didden Autoranger II, for example) plus software comes in well under $1000. For a few hundred dollars more you can add a calibrated mic and a small amp. With this equipment in hand, a user can achieve measurements at the same level as the professional gear, the tradeoff being time for integration, setup, calibration, and inevitably troubleshooting. And for many, that's a bridge too far.

Intermediate solutions have appeared on the market, such as Prism Sounds' dScope series—now Spectral Measurement—coming in at between one-third and one-half the cost of gear from companies such as Audio Precision. These systems integrate software and hardware in the same seamless manner as the higher-priced gear, albeit with slightly lower (though still excellent) performance.

Still, the small lab and home testing niche is one with limited options, so Echo Test + Measurement has stepped in with its AIO Test System.

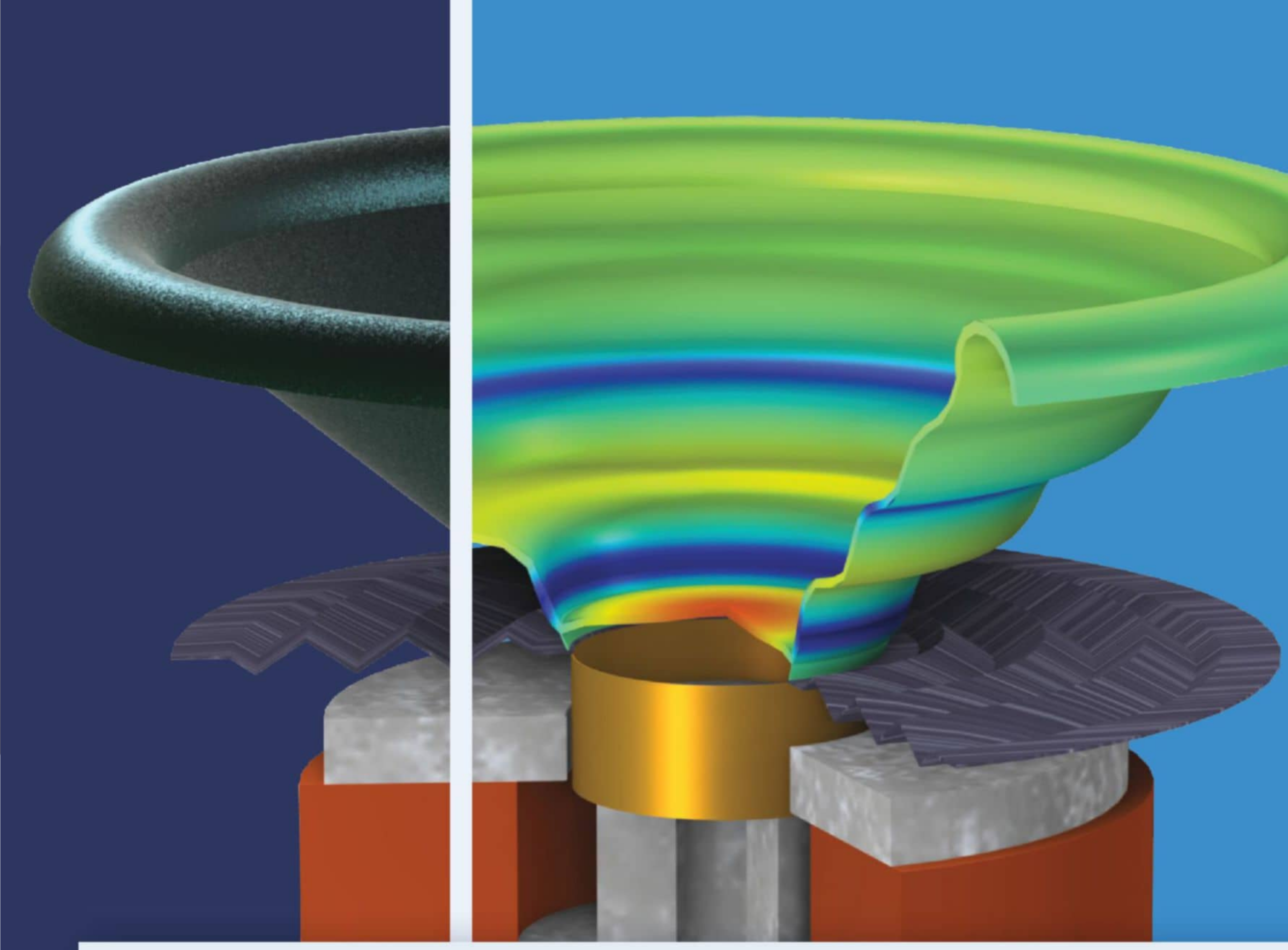

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Photo 1: The Echo AIO comprises two user-selectable modules mounted in a slim enclosure.

The Company

Echo Test + Measurement is the latest iteration of a Santa Barbara, CA-based company originally called Street Electronics, and later in the 1990s known as Echo Digital Audio. Its product lines have changed with the times—my first exposure to them was perhaps 15 years ago when its specialty was audio recording interfaces and external sound cards. I had a two-channel version of its AudioFire line, which I used as both a recording/playback interface and as part of a sound-card-based audio measurement setup with an older Macintosh computer.

The only problem was something that Echo couldn't control—the Firewire interface (equivalent to IEEE-1394) was standard for Apple products, until it wasn't. Apple changed over to a USB interface, and Firewire receded into history, along with a lot of now-obsolete development work. And the AudioFire went back into a box and is still collecting dust in my garage storage, a pity since its performance was excellent. I tried using an IEEE-1394 adapter to my Windows laptop, but that was clumsy and unreliable compared to the Firewire connection with my old Mac.

Echo made a lot of other great external cards (and at least one internal PCI card), but that's now a market with a lot of players and commoditization. However, the technologies integral to Echo's core competencies could also be applied to customizable commercial audio production test gear, and that brings us to the AIO Audio Test System.

The Product

Echo realized that test equipment for manufacturing lines that can be integrated into automated test systems was a natural use for its technology, and to customize it for wide use, its approach resembles mainframe oscilloscopes with various plug-ins. After making a large quantity of these systems for automated production line testing, Echo decided to try to sell it in the home lab and small commercial lab market, where the economies of scale brought the price down to a very affordable level.

The resulting AIO system is an enclosure with power supplies and support circuitry to which two modules may be added at the customer's request. The enclosure is a slim stackable unit with a pilot light on the front panel (**Photo 1**) and both power and I/O on the rear panel (**Photo 2**). The "mainframe" has the power and the USB interface, then space for two modules chosen by the user to customize the test system. Module options are:

AIO-A (Acoustic): The Acoustic module has four inputs that can be used for ICP/CCP microphone or as line inputs, and two amplifier outputs rated at 10W each into 8Ω. This module is suitable for basic speaker testing (frequency response, polar pattern).

AIO-C (Combo): This module is designed for logging environmental variables (temperature, pressure, humidity) rather than audio measurement.

AIO-H (Headphone): The Headphone module has two ICP/CCP mic inputs, suitable for embedded mics in artificial heads. There are two audio outputs with 3V and 125mA maximums, more than enough for the majority of headphones and headphone transducers. One is a conventional voltage output and the other has a 0R1 series resistor to generate a voltage proportional to the transducer current. The voltage developed across that resistor is then monitored by an internal floating differential amp. Additionally, the outputs are equipped with remote sensing (four wire or Kelvin connection) to allow accurate impedance measurement without having the impedance of cables contaminate the measurement.

AIO-L (Line): This module provides four single-ended line or ICP/CCP mic inputs with a maximum input range of 6.2V at an x1 gain setting, and two balanced floating outputs with a 2.0VRMS maximum output into a 600R load.

AIO-S (Speaker): The Speaker module is similar to the Acoustic module in that it has two 10W amplifiers built in, but has two inputs for line or ICP/CCP, and one of the two amp outputs has a 0R1 series resistor similar to the Headphone module to measure the speaker current. Like the Headphone module, the amp with a series resistor also has a four-wire connection for eliminating the effect of speaker cables in impedance measurement.

AIO-T (TDM): The TDM is a digital module useful for monitoring data streams in I^2S and DSP format.

Photo 2: The "mainframe" has the power and the USB interface, then space for two modules chosen by the user to customize the test system.

It offers eight channels, 192kHz sampling, and 24-bit or 32-bit operation.

Note that, unlike oscilloscope mainframes and plug-ins, the modules must be installed and configured at the factory and are not user-changeable. An AIO Test System with one module sells for $2750. Adding a second module bumps the price to $3250.

The microphone inputs are all ICP/CCP, that is, a single-ended constant current system with

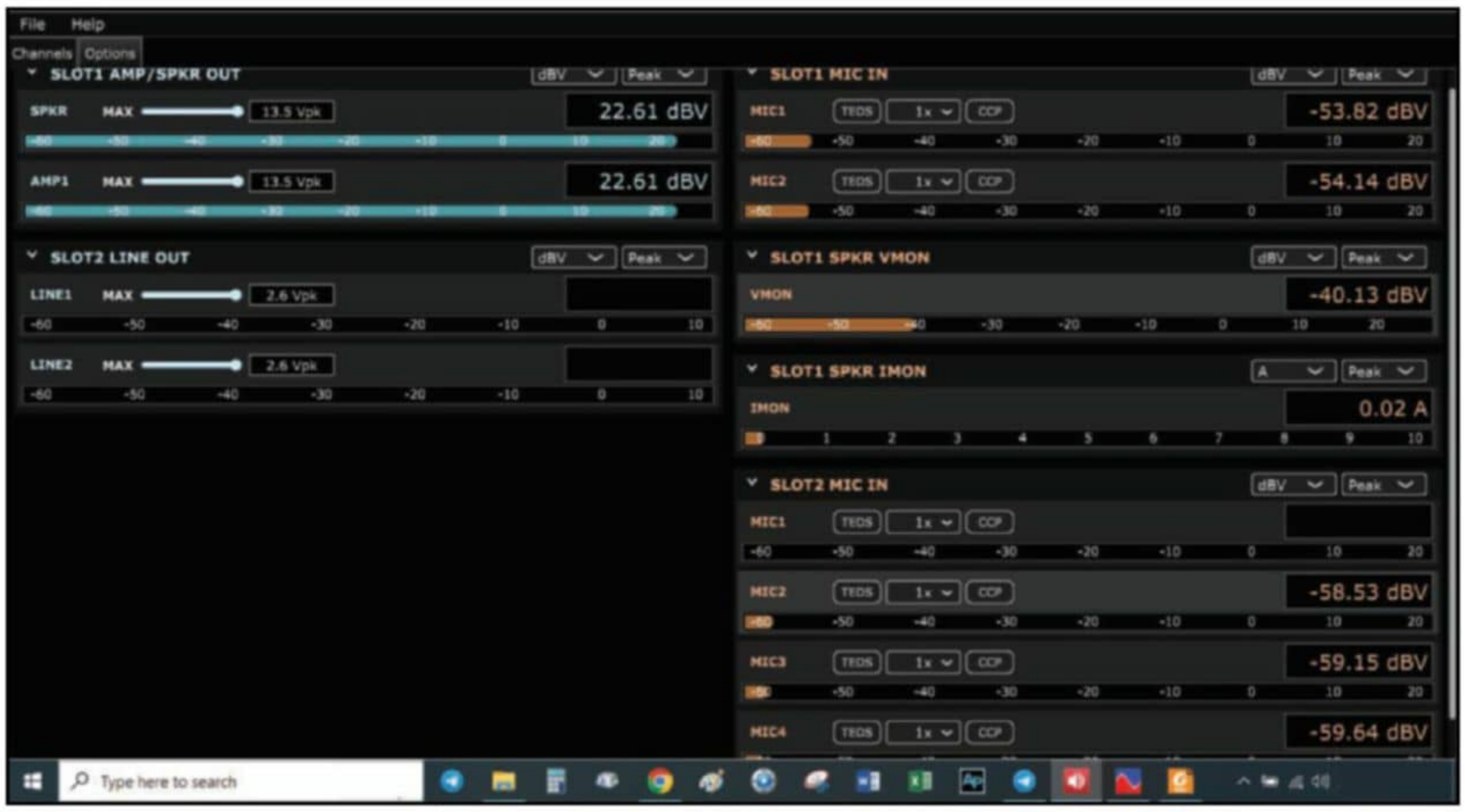

Figure 1: After installing the Echo drivers, plugging the unit into a USB port brings up the control screen.

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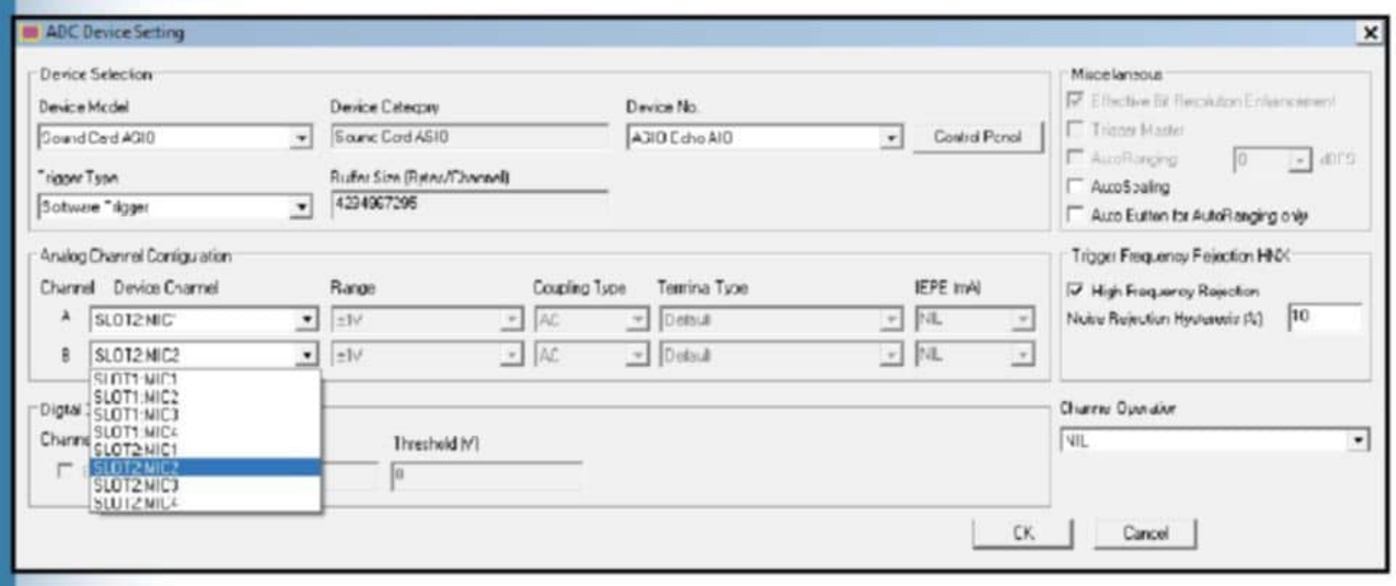


Figure 2: Test software like Virtins Multi Instrument shows the Echo as a sound card with selectable I/O.

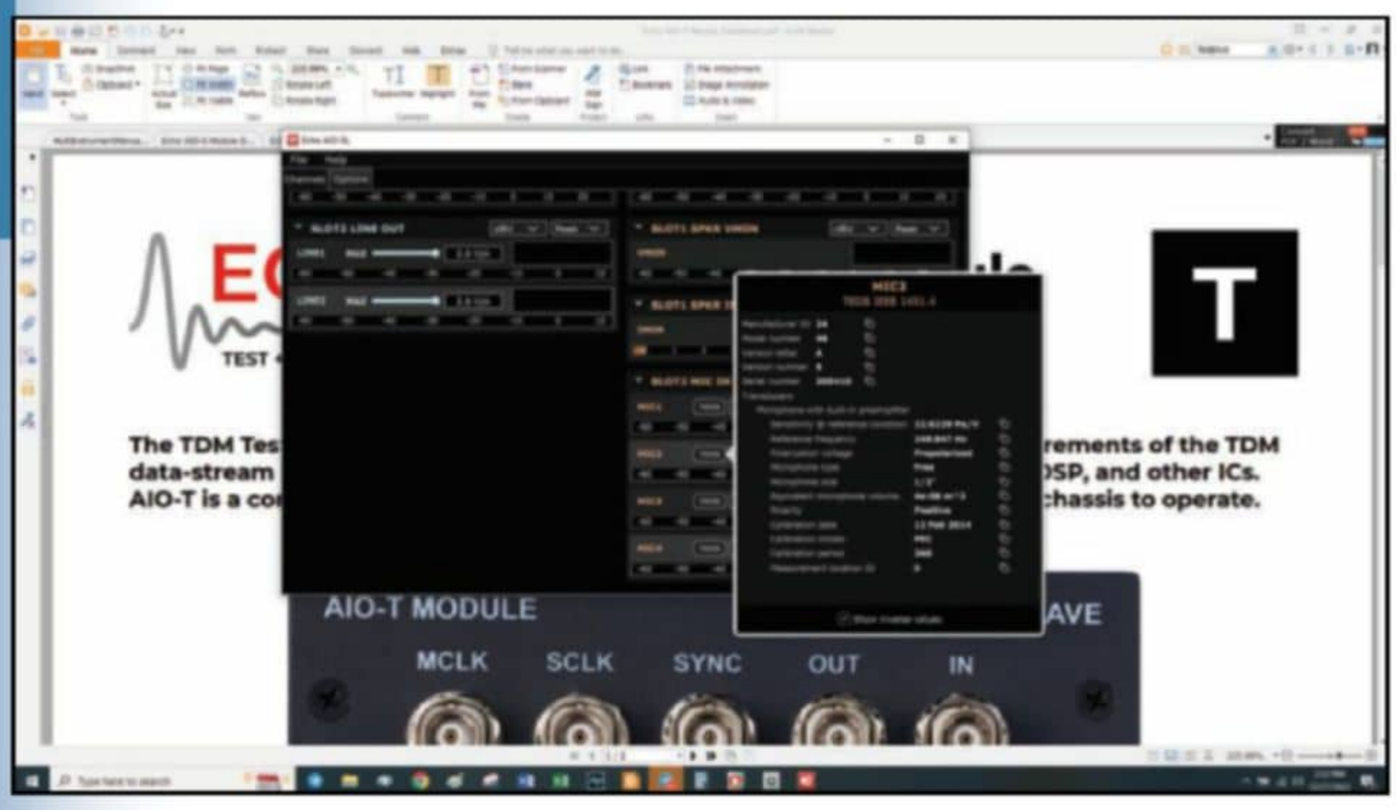


Figure 3: The ICP/CCP inputs of the Echo are TEDS-capable.

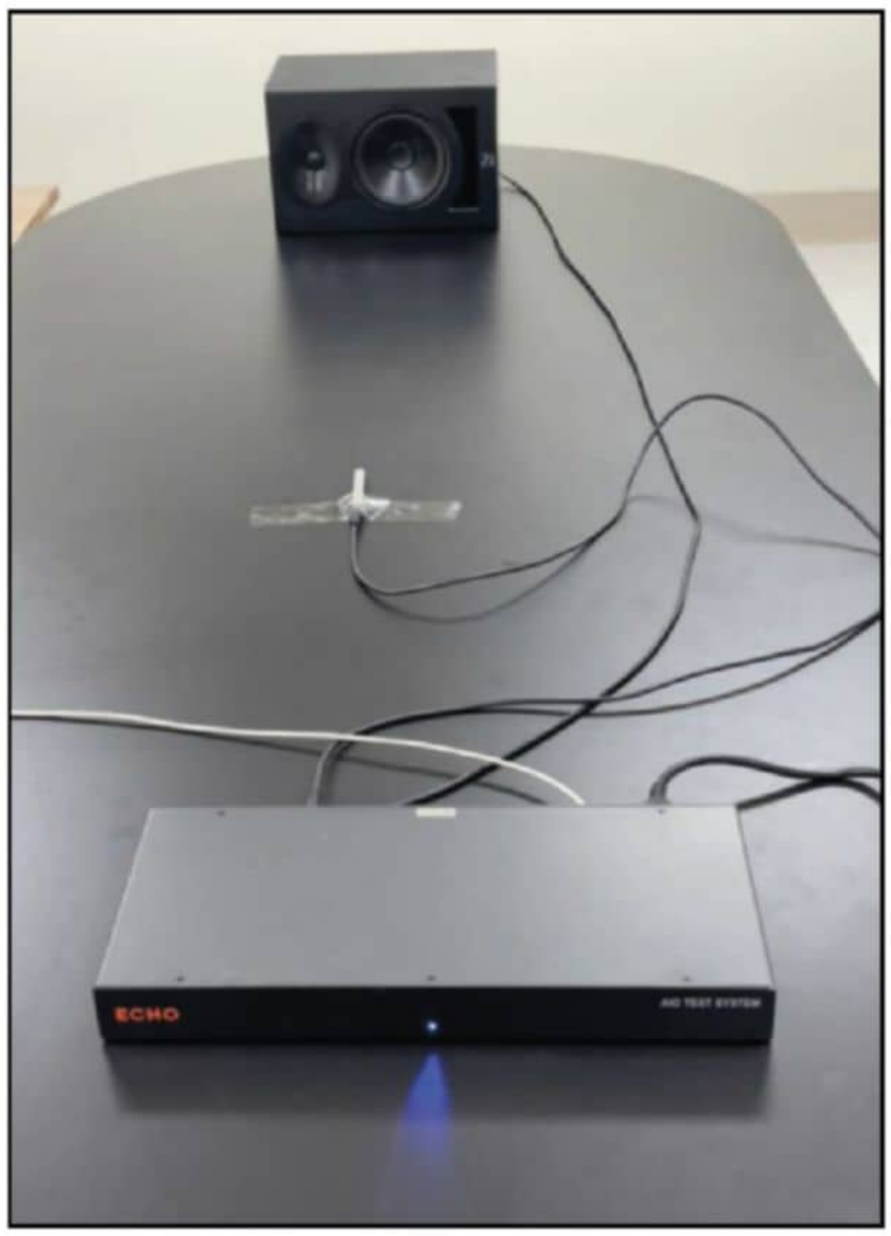


Photo 3: The AIO-L module was easy to set up for measuring an active speaker.

BNC connectors. I have written about this method previously (see Resources). It is common for industrial measurement, but less common in home and small commercial labs where balanced phantom power mics tend to predominate.

A major advantage of ICP/CCP mic systems is the ability of the interface to read calibration data directly from the mic via the Transducer Electronic Data Sheet (TEDS) or IEEE 1451 protocol. This can be a real timesaver compared to manual entry of sensitivities, especially when mics are being moved around between inputs. And if you have a lot of test mics around, without TEDS you also have the fun of finding the calibration sheets to get the numbers from… because the datasheet is built into the mic, TEDS is a lifesaver for absent-minded sorts like me.

Rather than use standard XLR or other common connectors in home or pro audio, Echo chose to use pluggable terminal blocks, which accommodate stripped wire ends that are screwed into a connector, which in turn is then snapped into a receptacle on the AIO plug-in. This requires the user to make adapter fittings for any standard connection, but has the advantage of being agnostic as to the type of connectors in DUTs. It also simplifies the mechanics of making connections because the user doesn't have to wangle screwdrivers into inconvenient locations, the connector block comes to him or her.

One excellent feature missing in nearly every other test system: the line inputs have an impedance of 1M. This means not only a reduced tendency to load down circuits being measured, but also that scope probes can be used. I hooked a standard 10× probe to one of the inputs and confirmed that it passed square waves at the correct voltage ratio.

The AIO in Use

Because of a particular set of measurements I wanted to do, the plug-in options I chose were the AIO-S and AIO-L. The AIO appears as a sound card when connected to your computer's USB port. The driver installation went smoothly, and when the AIO was connected, the control panel popped up (**Figure 1**). The control panel automatically populates according to the installed plug-ins. It allows the user to select sample rates, gains for each input (×1, ×10, ×100), turn on ICP/CCP power when an input is used for a mic rather than line, and set maximum levels for each output (this doesn't change the gain, so be aware that the output will hard limit at that maximum level). It should be noted that gain selection is manual—the AIO does not autorange, so the user needs to be aware of this when measuring some high dynamic range phenomena.

Additionally, the control panel enables the user to monitor input and output levels during test setup, in lieu of the measurement software package chosen. Speaking of which, during my evaluation, I used REW, ARTA, and Virtins Multi Instrument and it interfaced flawlessly with each of them, with their respective menus allowing the choice of any of the I/O ports on the AIO modules; **Figure 2** shows how the choices appear in the Virtins Multi Instrument device menu. According to Echo, the AIO also works with Audio Precision's APx Flex measurement software. I didn't have that on hand, but since the AIO appears as an ASIO device, I was able to simulate the last by running the APx500 software with the APx525 on my bench acting as a very expensive dongle, and choosing the AIO as either the input or output ASO device—after a few initial crashes and power down/restart, everything thereafter worked perfectly.

The control panel software can be addressed by command line or API if the user wants to automate a process. I note that the Virtins Multi Instrument software I used for many of my measurements can be programmed to access the control panel, run a test sequence, then output a QC check "yea or nay." So if, like me, you are averse to coding, automation can be easily customized with standard software.

My first quick check was configuring the Line and Acoustic modules' inputs as line input, then running a 1.000VRMS 1kHz sine wave from the APx525 into each of them. The voltage reading shown on the control panel was 1V for all inputs. Likewise, a 100mV signal at a 10× gain and a 10mV signal at 100× gain gave output readings of 1V, a tribute to the accuracy of calibration.

Since I didn't have an ICP/CCP mic, Echo was kind enough to loan me one, a GRAS 46AE. I used TEDS to retrieve the mic data (**Figure 3**), then I used a GRAS 42AB calibrator to check the sensitivity spec. This calibrator puts out 114dBSPL, so the mic output should be 10 times higher than the standard 1Pa/94dBSPL spec of 42.2mV/Pa. And at the indicated 0.424V, it certainly looks like all the calibrations line up very well.

With the AIO verified to be running properly, I pressed it into service doing some acoustic and electroacoustic measurements. First I used the L module to measure frequency response and

Fresh From the Bench

distortion of some active speakers I was modifying for use as desktop speakers (**Photo 3**). The line output of the L module was connected to the speakers, and the GRAS mic that Echo sent me was connected to the ICP/CCP inputs. The TEDS data was acquired, the calibration set, and then I used REW to get the acoustic data using log-frequency sweep (Farina chirp). Here's where the Echo was a major improvement over my usual APx525/1701 test system—no fan! Normally, I have to move the AP equipment and its associated lab computer to another room and run very long cables in order not to contaminate the measurement with the very significant fan noise from the two AP boxes. The Echo has no fan, is dead silent, and worked great just a few feet from the speaker.

Using the Echo, I obtained the graph shown in **Figure 4**, which looks pretty much like the same data I have been getting with the AP boxes, but in a fraction of the time since I didn't have to move equipment and run cables.

Likewise, I have been working on a project to map out actual voltage and current requirements for passive loudspeakers using real-world signals so that amplifiers can be designed to meet all use requirements for a particular application. The APx units are excellent for this since there's a current monitor built into the APx1701 acoustic interface, and in theory, you can acquire the voltage and current data separately, then graph them on an I vs. V Cartesian plot.

Unfortunately, the acquisition and storage of the data in such a way that the I and V can be lined up synchronously and at high resolution is a bit obscure (you'd think the Scope function would work, but it doesn't), and there is no provision in the APx software to create such a plot. Eventually, I found a workaround for the first issue, but the second requires all the data to be exported to Excel or similar package, then the graph created using the external software. I will attest that having Excel deal with multimillion cell spreadsheets causes spectacular slowing of my computer! Each plot took nearly an hour to acquire, process, and graph.

By contrast, with the Echo AIO-S module and Virtins Multi Instrument software, I had the measurement up, running, and graphing in minutes. **Figure 5** and **Figure 6** show I vs. V plots for a two-way passive speaker driven by the AIO-S's amplifiers at reasonably high volumes using two different sets of source material: pink noise and a 40 frequency multitone signal.

I'll be systematizing these data (and data acquired with music and speech signals) in a future article, but the point I wanted to make here is that the integration of the Echo AIO-S made this a fast and easy measurement instead of a struggle.

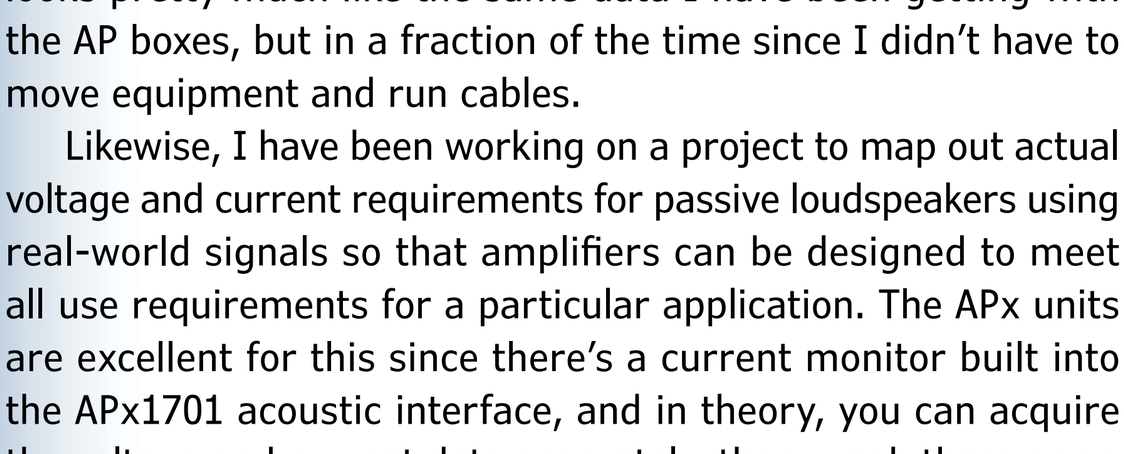


Figure 4: Using the AIO-L module and an ICP/CCP mic, I was getting good acoustic data from an active speaker within minutes.

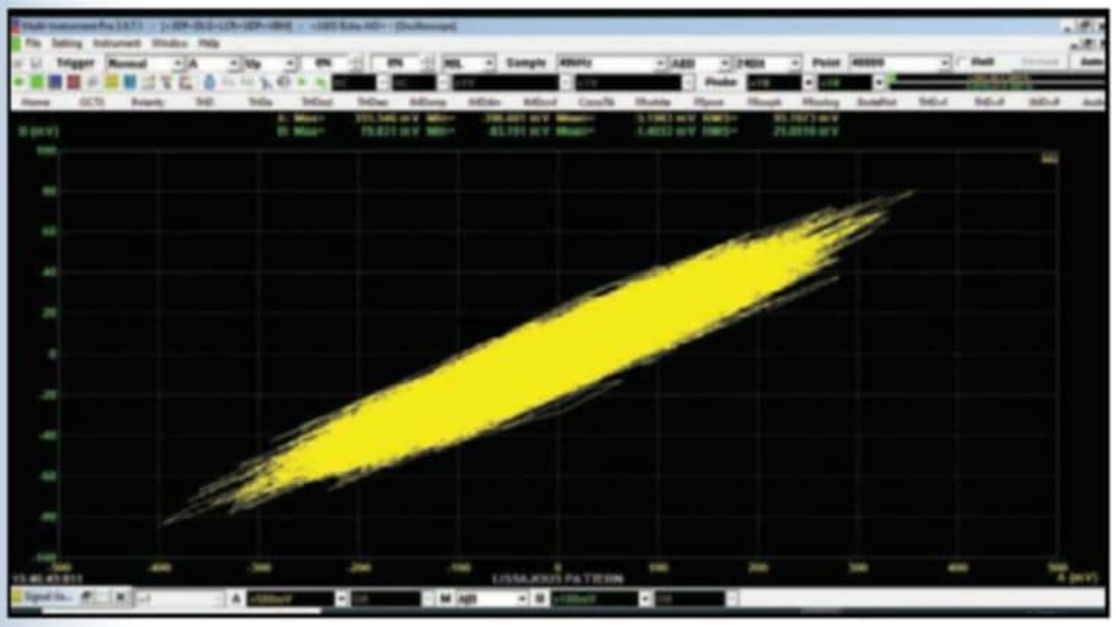


Figure 5: The Echo allowed me to get a current vs. voltage plot of loudspeaker drive using a pink noise test signal.

Echo AIO Test Measurements

I checked several aspects of the AIO's performance using an Audio Precision APx525. First, I attached 8R dummy loads to the Speaker module's power amplifiers. With no signal, the noise spectrum is shown in **Figure 7**. The ultrasonic and carrier noise is exceptionally low, with the worst component being under 3mV at about 300kHz. Next, I looked at the distortion performance of each of the modules. Starting with the AIO-L, I first looked at the

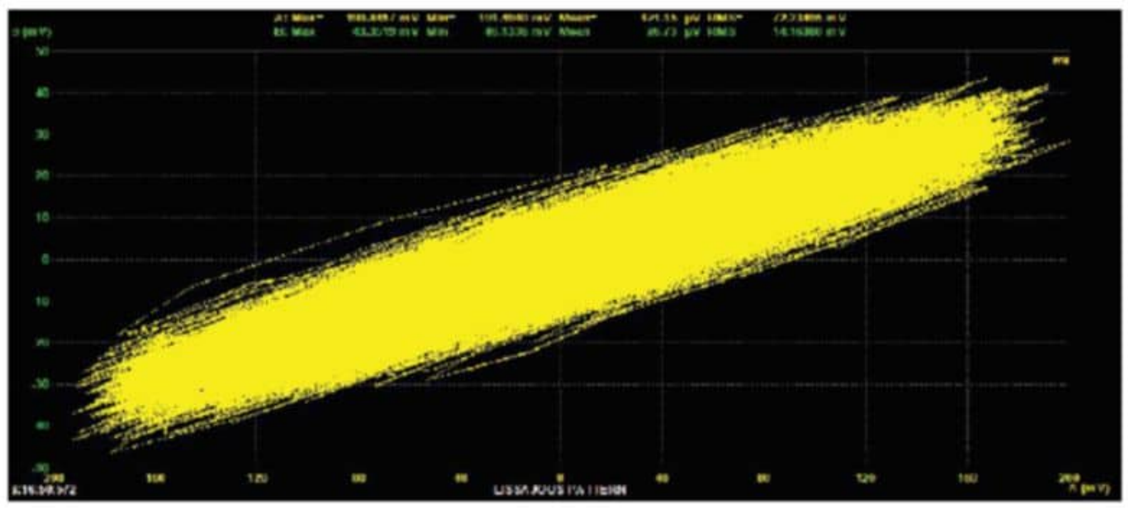


Figure 6: A multitone test signal gives a very different I vs. V plot.

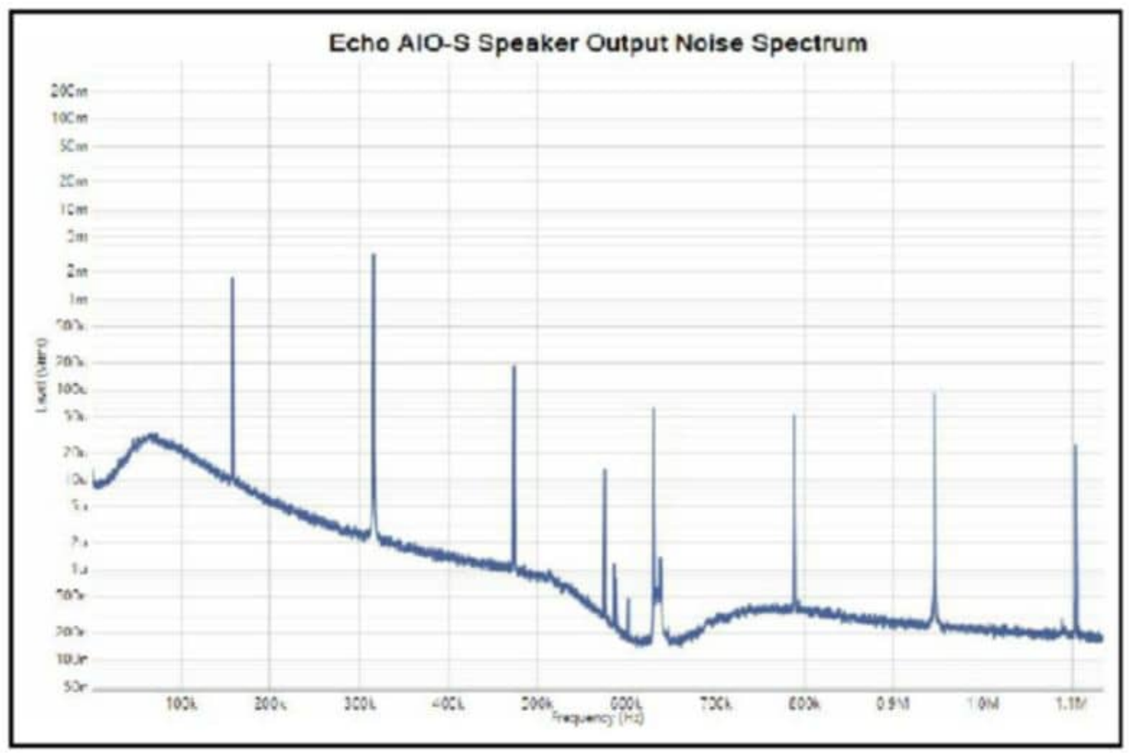


Figure 7: The output noise from the AIO-S power amplifiers is quite low in the ultrasonic range.

output distortion by addressing the input via ASIO, setting the level to -1dBFS and measuring the output in balanced mode. At that level, the output was 14.3V, so the full-scale voltage is about 16V. The distortion spectrum of a 1kHz sine wave is shown in **Figure 8**. The third harmonic is strongest at -108dB with respect to the fundamental. There's a lot of higher-order harmonics visible, albeit at extremely low levels. Raising the frequency to 10kHz at the same output level, I measured the distortion spectrum shown in **Figure 9**. The third harmonic has risen to about -103dB, and within the Shannon-Nyquist frequency limit, higher-order harmonics make themselves known. These are good results and good enough for most applications, but not at the level of current state-of-the-art electronics.

The input distortion results for a 100mV sine wave are shown in **Figure 10**. These were obtained by feeding a single-ended signal into the line inputs, then addressing the AIO output via ASIO. At a ×1 gain setting, absolutely no harmonics were visible. When the gain was increased to ×10, harmonics popped up out of the noise floor, with the second harmonic predominating at -105dB. I used a log frequency plot here to show that there's a noise rise at lower frequencies, and it scales with gain, suggesting its origin is the input stage.

I attached 8R dummy loads to the outputs of the power amplifiers in the S module, and applied a 1kHz sine wave. At 1V out, the distortion spectrum looked like **Figure 11**, with third harmonic predominant at -88dB. I increased the output to 2.83V (1W), then swept the frequency from 20Hz to 20kHz, obtaining the lower curve shown in **Figure 12**; the distortion stays at about 0.01%, but then starts rising at 1kHz.

The drop after 12kHz suggested that the anti-aliasing filters were suppressing the harmonics, so I reset the sample rate to 192kHz to capture the distortion at the top of the audio band. This is shown in the upper curve of Figure 12. The increased

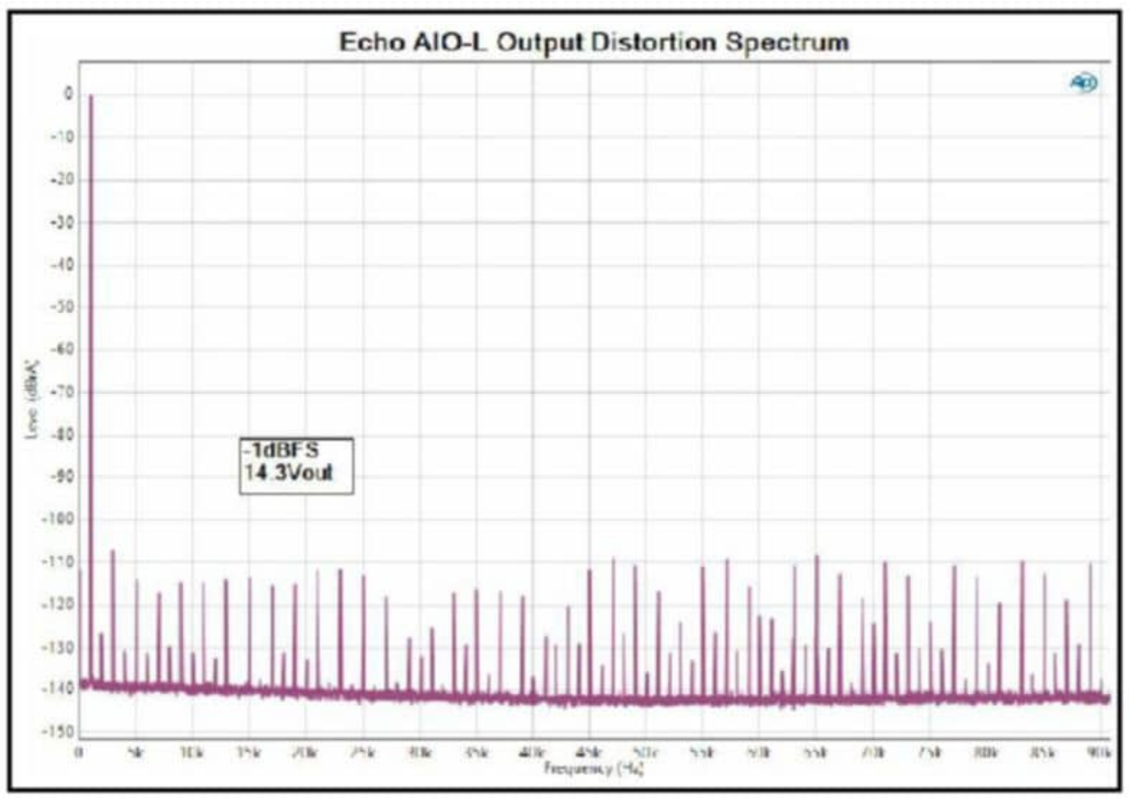


Figure 8: The output distortion of the AIO-L module at 1kHz and 14.3V shows a lot of high-order harmonics but at low level.

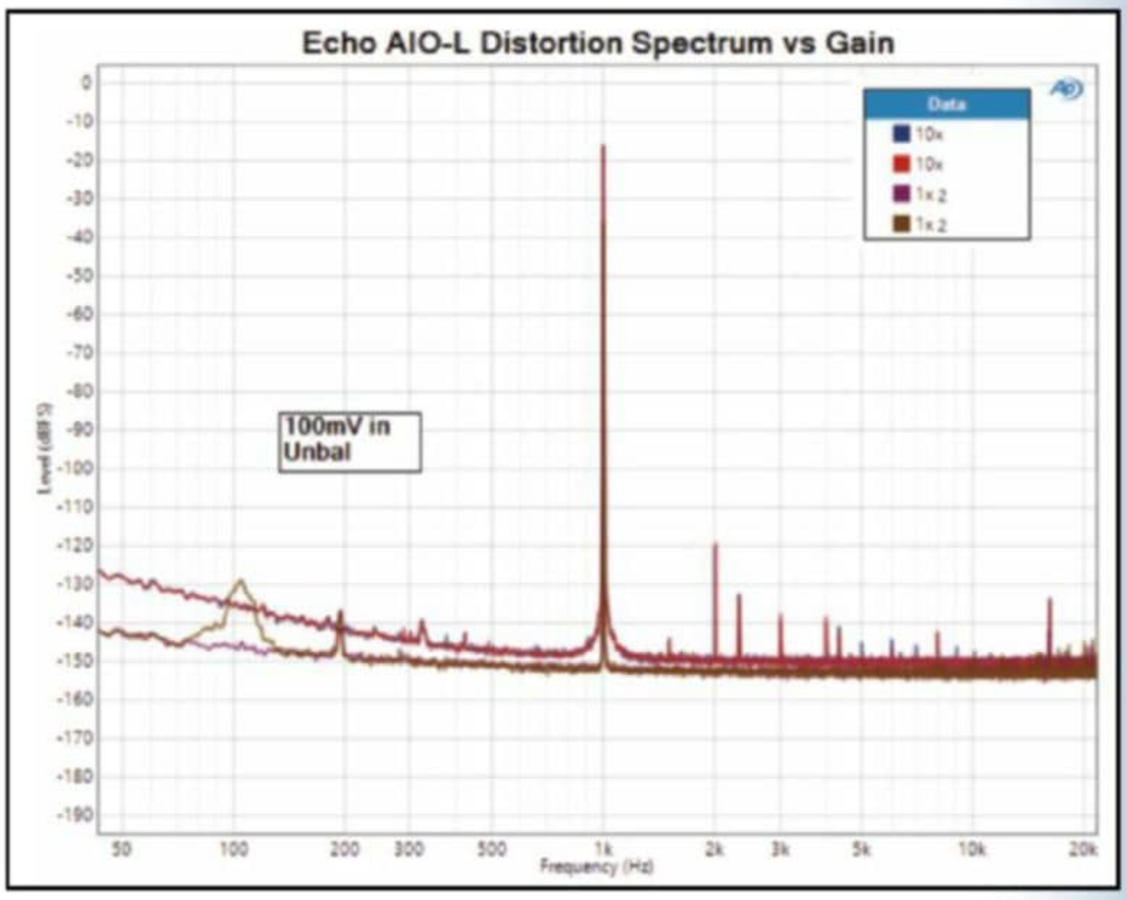


Figure 10: The input distortion of the AIO-L rises slightly with higher gain settings.

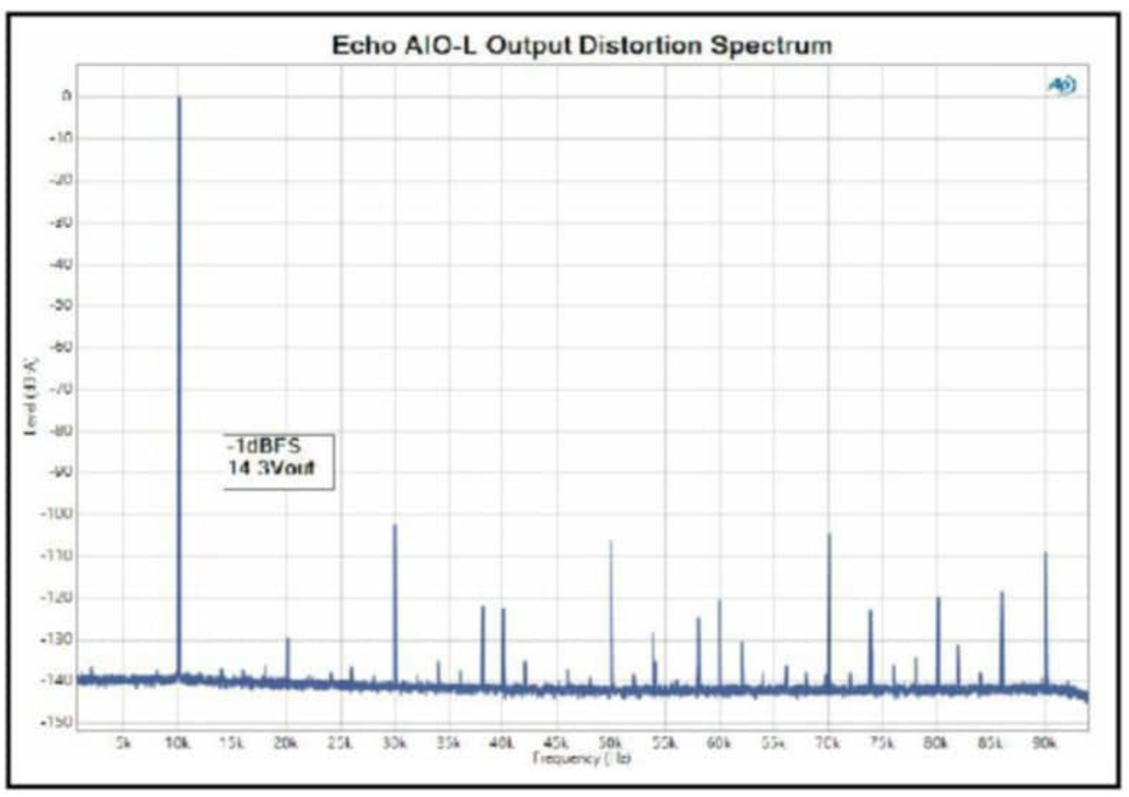


Figure 9: Raising the frequency to 10kHz shows no large distortion rise for the AIO-L module output even at near limiting voltage.

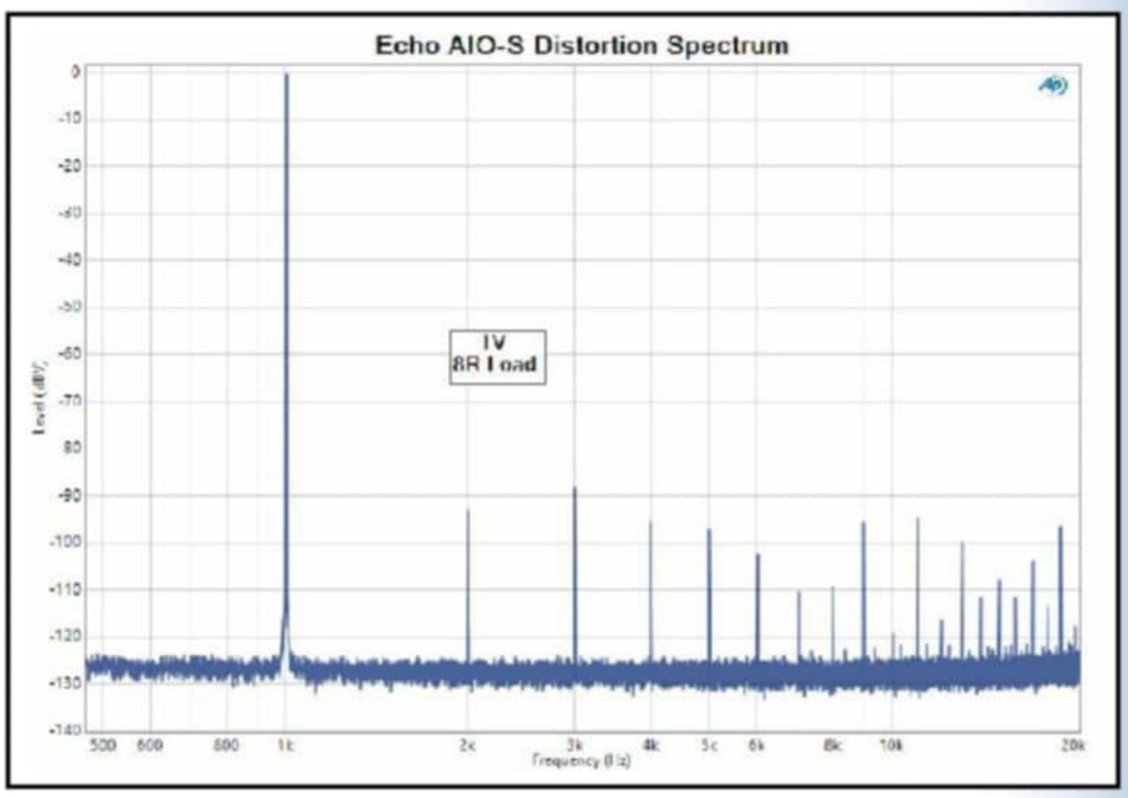


Figure 11: The output distortion spectrum of the AIO-S amplifiers at 1V and 1kHz is dominated by the third harmonic.

Fresh From the Bench

distortion below 1kHz at this sample rate is due to capturing more ultrasonic noise and won't impact normal loudspeaker measurements, but the high distortion (up to 3% at 20kHz) is due to the amps. Again, depending on the measurement you're trying to perform, this may or may not be a problem, but if I were (for example) measuring distortion performance of tweeters, I'd want to use the L module and an external amplifier.

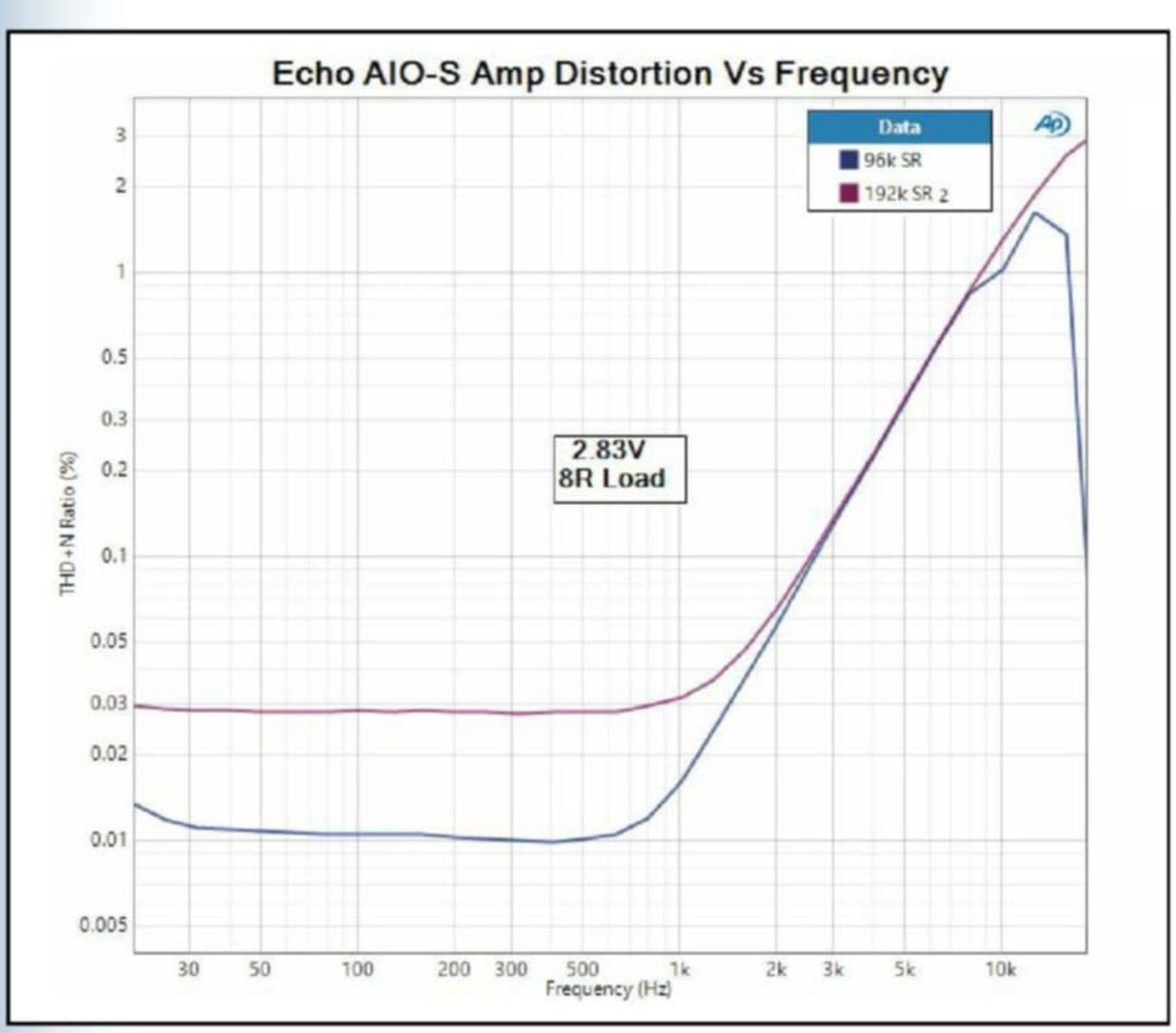
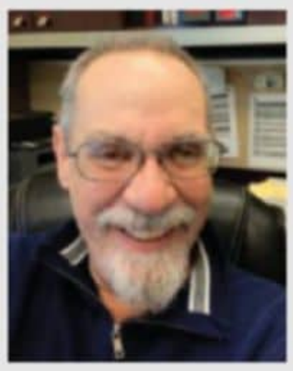


Figure 12: Distortion versus frequency for the AIO-S amplifier at 2.83V (1W) shows a very large increase at higher frequencies.

About the Author



Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently runs a technology consulting agency in western New York. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

Resources

J. Martins, "Echo Test + Measurement - Specifically Made for Audio Testing!" *audioXpress*, March 2022, https://audioxpress.com/article/echo-test-measurement-specifically-made-for-audio-testing

S. Yaniger, "Fresh From the Bench: Measurement Microphones from PCB Piezotronics," *audioXpress*, December 2016, https://audioxpress.com/files/attachment/2615

S. Yaniger, "The Virtins Multi-Instrument Software," *audioXpress*, March 2016,

S. Yaniger, "The Lab Bench of 2021: Virtins Multi Instrument Measurement Software," *audioXpress*, March 2021, https://audioxpress.com/article/the-lab-bench-of-2021-virtins-multi-instrument-measurement-software

Complaints, Cavils, and Squawks

Every silver lining has a cloud, and the AIO is no exception. Most of the issues that bothered me were a direct result of the AIO being originally designed for production lines rather than the test bench. For example, the connectors and power are all on the back of the unit rather than the front. This made attaching and removing cables—something rarely done in a manufacturing environment but constantly done in a home or small lab, more of a hassle than having all the connectors in front and visible. If you're using it in a more permanent bench setup (e.g., for QC tests), this is a non-issue.

And speaking of connectors, the terminal blocks are rather straightforward for adding whatever connectors are suitable. However, it is a bit inconvenient to make up the plugs and jacks, and they make a mess on the bench when all the I/O ports are in use. It would be nice to have a few standard connection options (e.g., combo XLR/TRS and phono plugs built into the unit), or offer a patch panel as a plug-in option.

Although ICP/CCP mics are common in large labs and commercial facilities, I think my exclusive use of balanced phantom power mics might be more common for small labs. It would be nice to have that option; because ICP/CCP is the only mic input option, I borrowed a mic from Echo for this evaluation. (It was an extremely nice one, thanks, guys!)

Conclusions

The Echo AIO has some great things going for it—compact integrated design, physically quiet (no fan noise!), extremely easy to configure, and seamless compatibility with every piece of measurement software I threw its way. Measurements that were massively painful using other test systems were trivially done with the AIO. The ability to use standard scope probes is also a blessing. For a small manufacturer, it would be a great addition to outgoing QC and characterization.

The shortcomings about which I previously complained are a legacy of the original design intent and are all relatively easy to deal with. The AIO with the modules I ordered does not have state-of-the art distortion performance, but it's good enough for most applications.

Where it really showed its stuff was in acoustic measurement, where the lack of fan noise made setup and cabling much easier, and the versatility enabled novel measurements that I could not do easily with APx or CLIO. At its relatively low price and with its ability to be customized, small labs and manufacturers will find great value in the Echo AIO.

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Market Update—Test & Measurement

Modular Evolution in Audio Testing

By
J. Martins

The audio industry continues to benefit from significant progress in advanced software as well as modular hardware solutions. This overview of the latest audio and industry-related test and measurement solutions identifies market trends for this critical segment.

The previous months were unusual for the test and measurement sector and specifically for audio and acoustics testing. With the effects of the global pandemic subsiding, the return of traveling and getting back to the production floors, test and measurement manufacturers are busier than ever in the field. Except, access to factories in China for visitors from other countries is not yet fully available, which continues to push the trend for remote access and supervision, with test equipment required to be connected and sharing data on the cloud. Something that audio test and measurement manufacturers are working on but have not yet fully implemented.

Still, as a result of the additional time for R&D activities available through the pandemic, the results of intensive work on the software front are now clearly visible, with many new features and frequent major updates, while hardware remained on hold given that it requires a more collaborative effort that was not completely available during the pandemic. But more important, the effects of the supply chain disruptions continue to be felt, causing the postponement and delays of product launches—sometimes even products that had been announced and pre-orders accepted.

Available or Not

A great example of the ruthless supply chain challenges felt across the industry are perfectly illustrated by QuantAsylum, which was forced to discontinue previous designs due to key components no longer being available, including AD/DAs from AKM. Since 2012, the company founded by Matt Taylor has been providing audio test and measurement solutions to companies around the world, and with each iteration, the capability and programmability of its audio analyzers continued to grow.

In January 2022, the Seattle, WA-based company was planning to launch the QA403, its fourth-generation audio analyzer that resulted from the need to redesign the QA402 and workaround the discontinued components. The QA403 moved to better-performing AD/DAs by ESS Technology and, as the project evolved, the overall design performance promised to be a substantial leap beyond the previous model. Yet, 2022 came to an end and very few units shipped to customers.

After all the redesign effort involved, in 2022 the company was unable to sustain production of the QA403 due to unexpected shortages of common components from Texas Instruments for which no alternatives existed. This led Matt Taylor to consider another redesign that resulted in a planned QA404 model. More recently, the component supply situation improved sightly, and the company is planning to resume production of the QA403, while offering the new QA404 as a better performing (THD) option for just $50 more. Both units should be shipping by Q2 2023.

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Market Update—Test & Measurement

While not intended to compete with Audio Precision or top of the range scientific instruments, the QuantAsylum QA403 is a professional-level instrument that is perfect for a developer's or enthusiast's bench, as well as for the requirements of manufacturing lines. And it comes with the advantage that QuantAsylum's software can be easily programmed and customized for such applications.

The company also offers the possibility to pair a QA403/QA404 with any other of its interface and input systems, such as the QA460 Transducers driver and a pair of QA471 mic preamps, all in a custom design 2U rack mount plate, that together yields a powerful speaker tester at a very competitive price.

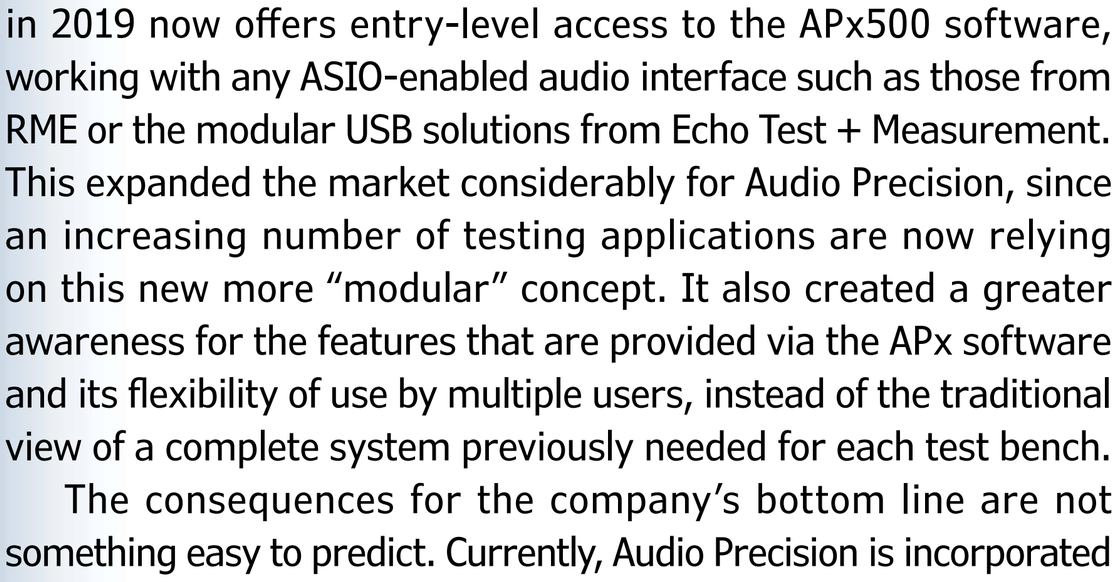
QuantAsylum is a small business compared with other established audio test and measurement manufacturers, but the volumes the company regularly reports in its production updates are great indicators of how the industry evolved, from a time where there was basically a handful of traditional suppliers.

More Complete Solutions

Audio Precision (AP) continues to sell its APx515B and APx517B audio analyzer for production test applications, together with the APx52xB and the APx555B Series for more demanding R&D applications, and its more recent APx1701 transducer test interface was certainly well received by the audio industry.

Meanwhile, the launch of the APx500 Flex Audio Analyzer in 2019 now offers entry-level access to the APx500 software, working with any ASIO-enabled audio interface such as those from RME or the modular USB solutions from Echo Test + Measurement. This expanded the market considerably for Audio Precision, since an increasing number of testing applications are now relying on this new more "modular" concept. It also created a greater awareness for the features that are provided via the APx software and its flexibility of use by multiple users, instead of the traditional view of a complete system previously needed for each test bench.

The consequences for the company's bottom line are not something easy to predict. Currently, Audio Precision is incorporated as part of the measurement-focused business group established by Battery Ventures, together with GRAS Sound & Vibration, and imc Test & Measurement, under the umbrella brand of Axiometrix Solutions. Supporting a global customer base in the test and measurement space for the most technology-intensive industries including electronics, aerospace, automotive, and audiology, Axiometrix Solutions is building a systems approach, which for audio applications implies the bundling of AP and GRAS products.



Audio Precision is offering its APx500 audio analyzers and measurement software with built-in support for the GRAS SysCheck2 microphone calibration, creating convenient integrated solutions for the audio industry.

An example is the expansion of Transducer Electronic Data Sheet (TEDS) calibration functionality to all GRAS equipment and integration with Audio Precision systems to ensure reliable data and to ease all aspects of installation and test platform setup.

This certainly increases the appeal for AP products at the higher level. As Axiometrix Solutions detailed in its original corporate mission, "to use application knowledge, innovative engineering, and operational expertise to enable engineers, technicians, and researchers to move from challenge to insight."

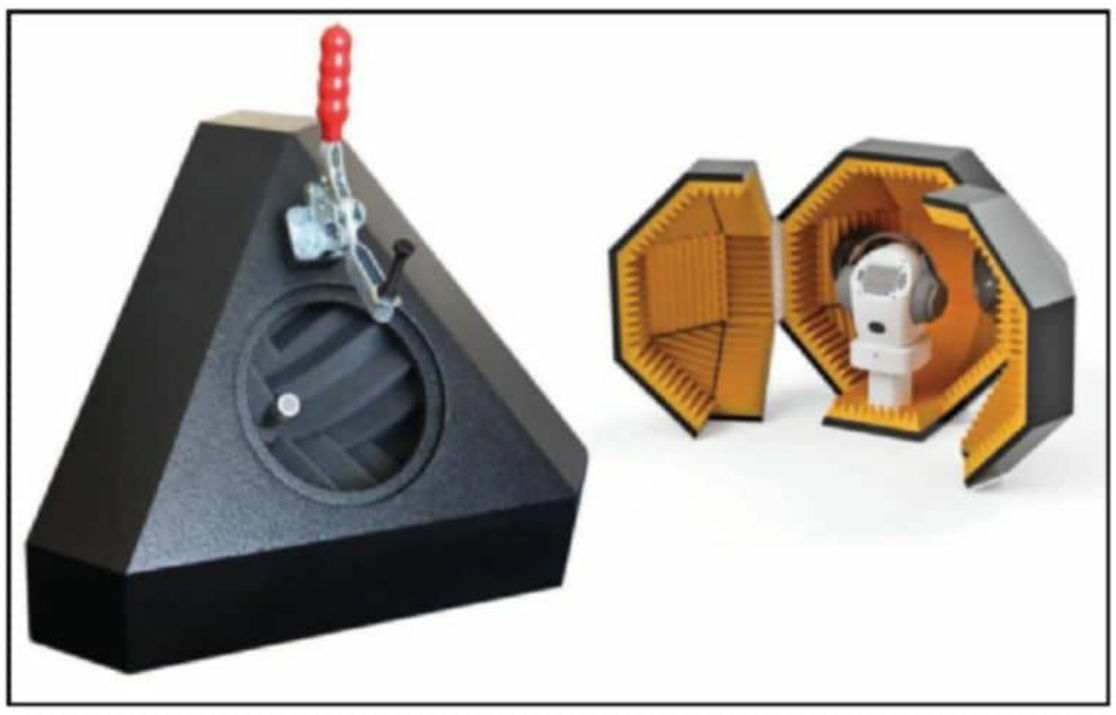
But this approach of selling more application-oriented solutions, rather than individual equipment is somewhat challenged by the modular, interoperable solutions that are increasingly available and offer greater flexibility.

Echo Test + Measurement expanded this flexibility one step further and offers a complete range of interface and application-oriented modules to its USB-based audio test and measurement systems, which all work like standard sound cards and are compatible with most audio test software. A concept that enabled Echo to a rise to a leading position in systems for manufacturing, quality assurance/quality control, component and sub-assembly verification, and secondary test bench applications.

And yet, like many other companies, Echo didn't escape the supply chain hurdles of the past two years. As delivery times have extended to multiple months on new test and measurement hardware from many manufacturers, Echo was also forced to delay the launch of many of its modules that it planned to offer in 2022. As availability expands in 2023, there's a clear sign that the company could quickly expand its presence in the industry. It will be interesting to see how existing software-centric test and measurement companies will gain from the integration with Echo modular hardware.

Modularity

This increasing software and hardware modularity could pave the way to a new approach in audio test and measurement, evolving away from the all-in-one audio analyzer designs, directly into completely customizable systems. For general electronics instrumentation this was already happening, with more USB-based instruments exploring the increasing computation power of PC/Macs and even mobile platforms, such as tablets. The evolution to USB-C and the ability to power instruments via the USB bus is certain to boost that approach even more for affordable bench instruments. With USB Power Delivery 3.1 published in May 2021, there are now fixed voltages of 28V (above 100W), 36V (above 140W) and 48V (above 180W) joined by a new adjustable voltage mode enabling a range from 15V to one of three maximum voltages (28V, 36V, or 48V) depending on the available power allowing the device being powered to request specific voltages to a 100mV resolution. Add to this the availability of USB hubs that can easily power two 4K monitors and connect/power multiple



A Hill Acoustics-developed Tetrahedral Test Chamber (TTC) and an example of a new Polar Test Chamber adapted for headphone tuning and ANC evaluation with a built-in ZUMI Systems HP70 testing unit, now available from Spectral Measurement and DB Enterprises.

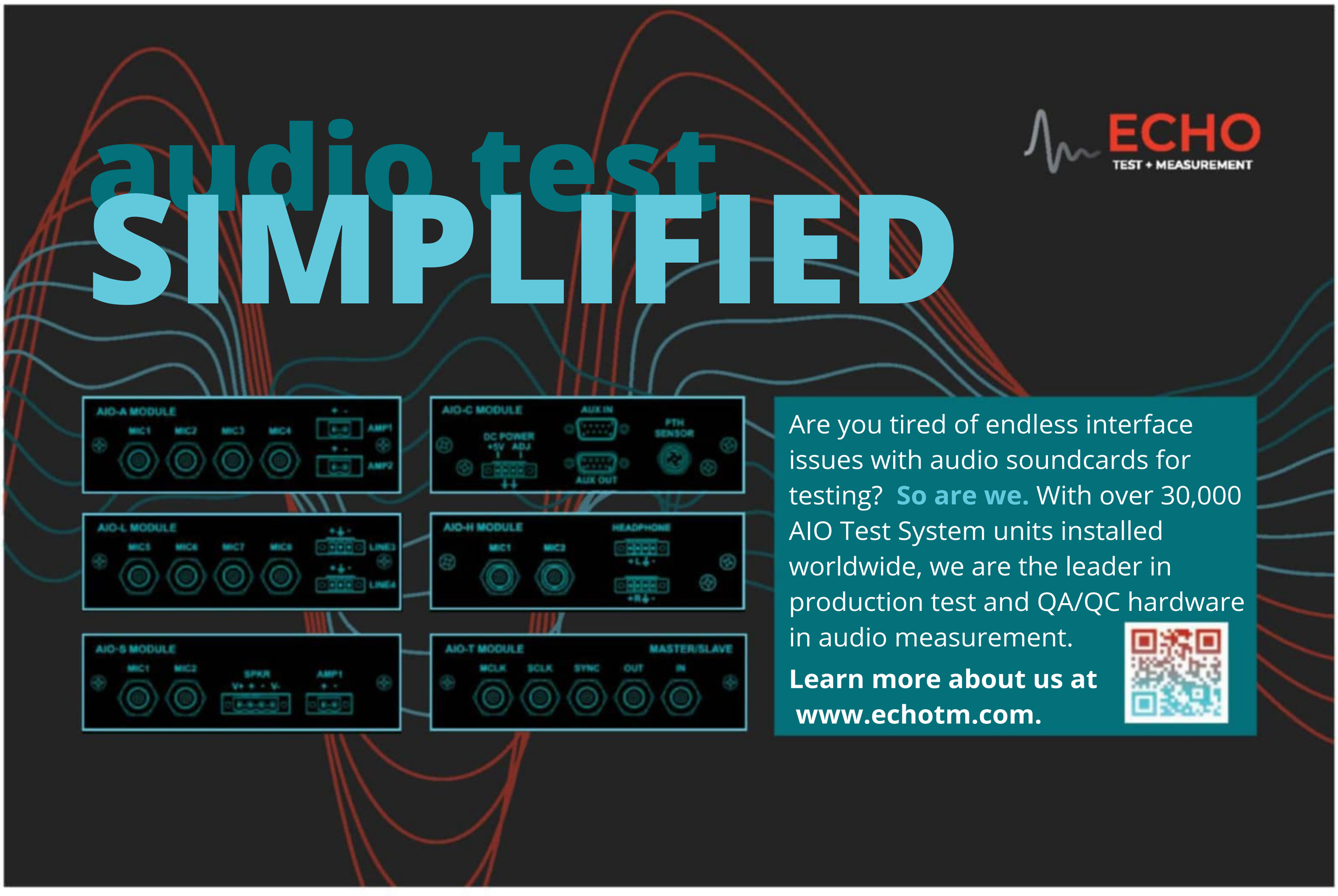
devices, and it becomes possible to create very powerful and versatile bench setups.

Some companies that had already anticipated this trend and were moving into that more "modular" and connected direction were unfortunately not able to make much progress when the pandemic put a hold on existing plans.

Spectral Measurement is one example of a company that was hit hard by the pandemic, right after it had announced a new structure and focused market strategy. In 2019, Spectral Measurement, the former Test & Measurement division of Prism Sound, was created as a company focused exclusively on audio test and measurement solutions, and separated from Prism Sound's Music and Post Production division. Graham Boswell, Prism Sound founder and now managing director of Spectral Measurement, had great plans for the company that offers the innovative dScope Series III Audio Analyzer, and developed innovative, high-quality audio test and measurement solutions since 1987. With a healthy order book in 2019, Spectral also introduced the new dScope M1, a new compact and much more affordable hardware interface that was compatible with its existing audio test and measurement software for audio electronics and transducer testing applications.

That same year, Spectral Measurement also started to offer the range of innovative loudspeaker tetrahedral test chambers that replace large and expensive anechoic chambers with small and inexpensive calibrated enclosures. These new products, manufactured under license from Hill Acoustics, are aimed at developers, manufacturers, and maintenance providers in audio electronics and electro-acoustics.

The pandemic placed the volume production of the new dScope M1 on hold—expected to resume now in 2023—and Spectral



Market Update—Test & Measurement

Data from an audio analyzer is uploaded to The Lyceum platform and graphed to visualize performance. Manufacturing data is uploaded according to user-established parameters allowing a pass/fail rate to be calculated, as an example.

Measurement has for now focused on extending its partnerships for the test chambers, including an integration with Zumi Systems for an acoustic test and measurement system for running and testing ANC performance and production quality control of headphones/earbuds.

Spectral also introduced new test chamber models available in various sizes, as well as non-tetrahedral custom designs that enable, for example, easy 360-degree polar response measurement. In many of those systems, Spectral will offer the dScope M1 as an hardware interface option, and is planning on also packaging other hardware interface solutions from manufacturers such as Echo Test + Measurement.

Another major trend that I believe is important to consider is the increasing importance of software connected services adding to test and measurement (and simulation) systems. A great example is the startup founded by Joshua Levy, a hardware engineer with more than 10 years of experience working with the products divisions of Amazon and Facebook. He created an online collaborative software, The Lyceum (www.thelyceum.io), which facilitates hardware system engineering research and development and manufacturing by automating data-intensive tasks, keeping a track record of historical technical data, and offering tools for online group work. This platform, not only facilitates hardware system engineering team efforts, but can also extend to the production floor and manufacturing statistics.

According to Joshua Levy, who has already partnered with existing audio test and measurement companies, The Lyceum offers an extra layer of convenience for the industry, accelerating analysis of large amounts of data and integrating team collaboration.

Currently, The Lyceum can functionally service audio systems engineers in the product development process. It ingests measurement data from software such as Listen Soundcheck or Audio Precision APx, and provides users the ability to graph/visualize, calculate statistics, and generate yield numbers from a manufacturing build.

For many companies that lack the resources to build and customize collaboration tools—never mind supporting the safety requirements of data systems on the cloud—integration with such platforms is a way to accelerate the implementation of connected interfaces, adding value to the services provided to an existing user base.

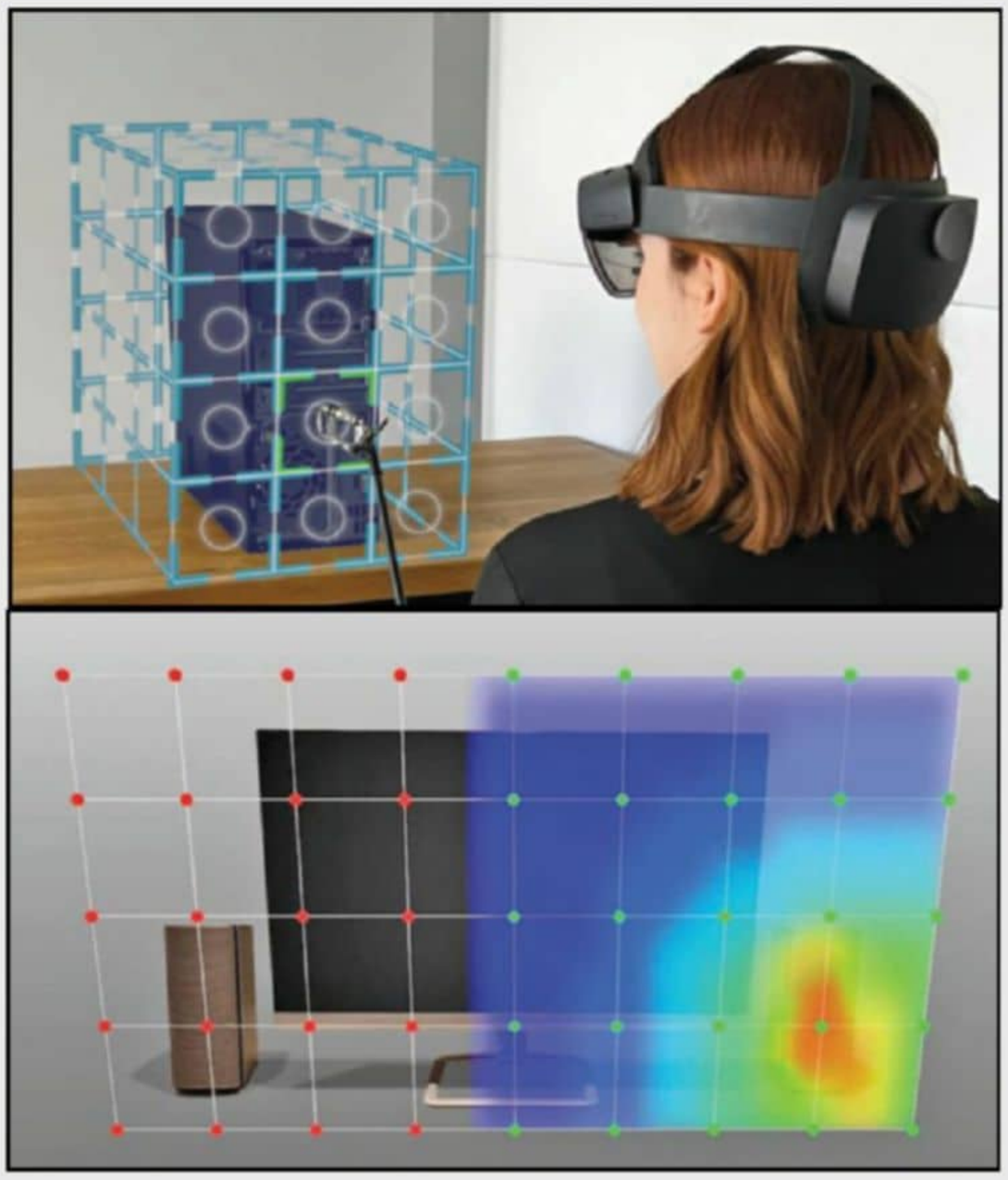
HEAD acoustics

Artemis Suite Evolution with HoloMetrix AR Interface

HEAD acoustics is one of the world's leading companies offering holistic sound and vibration analysis solutions, and its Artemis Suite is a software platform for measuring and analyzing sound and vibration in applications such as product optimization and environmental noise. The modular software solution combines several tools in a standardized environment, enabling applications for recording sound, vibration, and other variables such as temperature, stresses and torques, recording and decoding bus data such as CAN, CAN-FD and FlexRay, documenting recordings with user-defined metadata, analyzing the recordings using varied algorithms and sophisticated representation tools, and finally implementing standardized tests for validation, among many of its tools.

The latest major update, Artemis Suite Release 14, introduces new methods for DIN EN ISO 9614-1 measurements, moving away from manual measurement and measurement preparation to an efficient virtual solution with augmented reality (AR) glasses that enable visualization of required sound intensity measurements. This was made possible by a new interface to HoloMetrix's software solution Sound HUB.

With this new interface in Artemis Suite, users can explore augmented reality to guide even inexperienced users safely through the entire measurement procedure, and enabling visualization of the measurement progress as well as the measurement results. In combination with the sound intensity tool, Artemis Suite users can now measure sound intensities to determine the sound power or create sound intensity maps for quick source localization using AR glasses and a virtual, standard-compliant measurement grid superimposed on the real test object. This saves time and costs, completely eliminating the time-consuming physical measurement grids or the manual measurement of the probe position otherwise



The Frankfurt, Germany-based startup, HoloMetrix GmbH, has developed a software platform for acoustic measurements based on Augmented Reality, which projects the virtual measurement grid in three dimensions in the actual measurement environment. Working together with HoloMetrix, HEAD acoustics has now developed an Augmented Reality interface to its Artemis Suite software, allowing a simplified setup for sound power and sound intensity measurements according to the ISO 9614 standard.

required for measurements. Instead, users can position the measuring probe simply, safely and precisely with the aid of the virtual measuring grid.

For assessment of sound power, according to the relevant ISO 3740 series standards, which are based on sound pressure measurements, HEAD acoustics introduced separate analyses for the K1 background noise correction and the K2 environmental correction. This allows direct automation of these corrections during sound power measurements, increasing the efficiency and quality of results.

The new Artemis Suite Release 14 also introduces improved structural analysis tools that make it easier and more efficient to import data from other systems, or correcting errors. To make the quality or results even better, the MAC visualization with adjustable limit values is now realized by the modal analysis software directly during the calculation. And users can export vibration modes to Excel to prepare reports and enable further evaluations and analyzes.

In the field of psychoacoustics, HEAD acoustics has introduced innovations such as additional annoyance weighting for tone content (hearing model) and specific tone content (hearing model). This allows additional weighting of results for frequencies above 1kHz that people find particularly annoying. In addition, users can now calculate loudness analyzes using the Moore-Glasberg-Schlittenlacher method. The underlying standard ISO 532-3 has been published as a draft, and the calculation is already integrated in Artemis Suite Release 14.

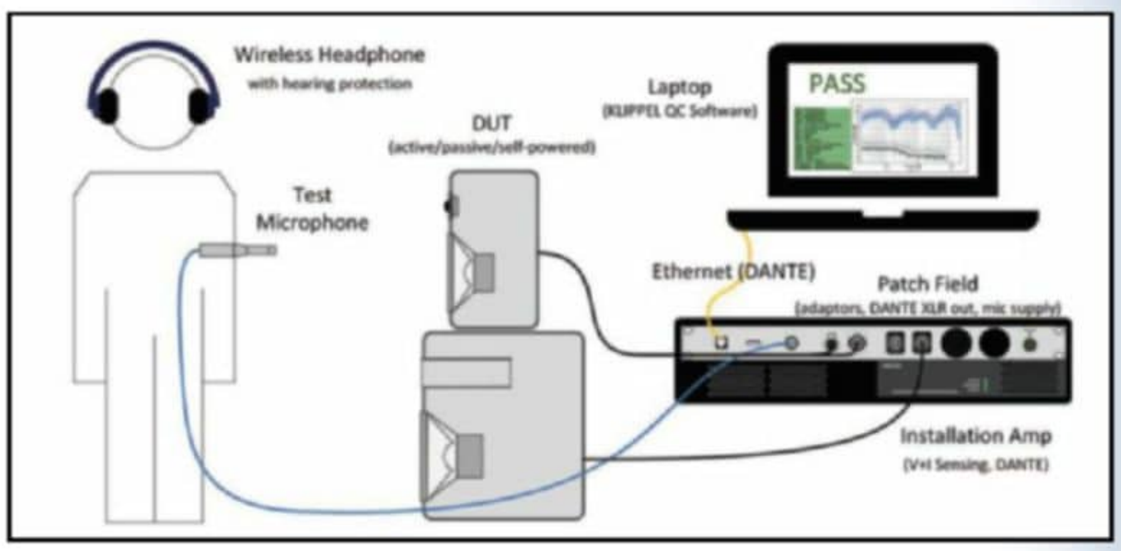
www.holo-metrix.com
www.head-acoustics.com

Klippel

Improvements to dB-Lab and QC Analyzer System Software

As always, the Klippel engineering team continuously introduces the results of its latest R&D efforts and customer feedback to improve its solutions. In 2022, Klippel released several software updates for the Klippel Analyzer System with various new features for both R&D and QC applications. The updates are free of charge for all users running dB-Lab 212 or QC7.

The Klippel Analyzer System now offers an improved Time Frequency Analysis (TFA) module showing the new wavelet transform of chirp response and filtered playback feature. As a dedicated post-processing tool for the analysis of waveforms in time and frequency domain, the module can import any signal or impulse response generated by the Klippel Analyzer System or imported from a wave file. This powerful post-processing tool now also provides an interactive band-pass filter and playback function for subjective distortion analysis and diagnostics, in addition to the spectrogram analysis options based on



Mobile quality control test setup for rental companies based on QC Stand-Alone Software and smart amplifier with Dante interface.

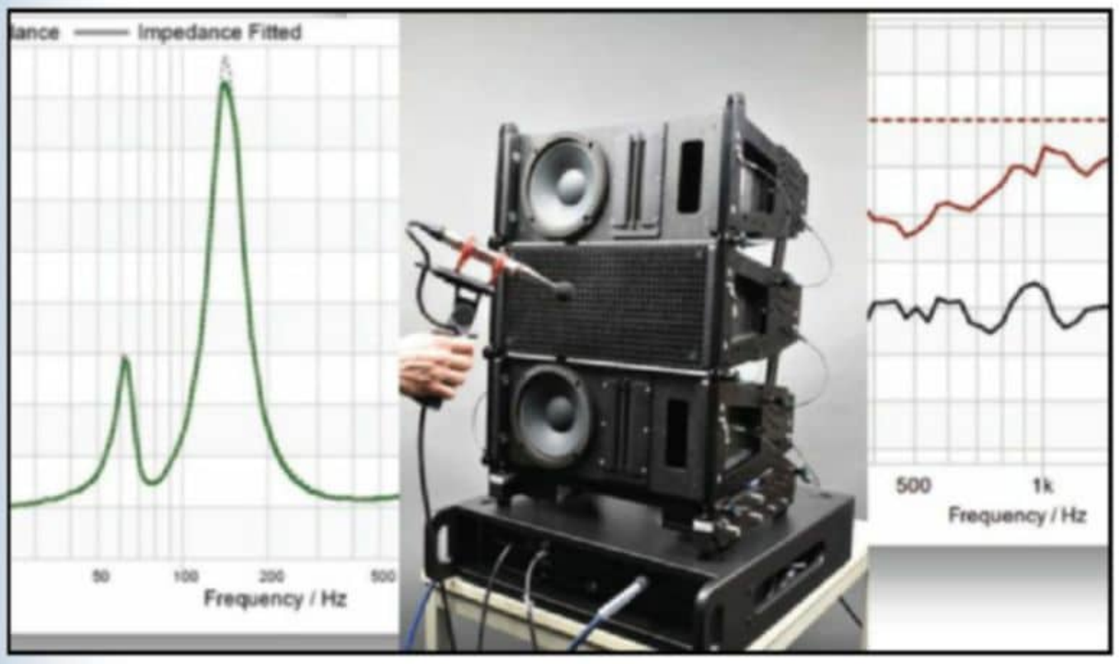
Market Update—Test & Measurement

Short-Time Fourier Transform (STFT), wavelet transform, or auditory filter banks with superior time resolution.

Another new feature is the Documentation (DOC) module for any dB-Lab operation (R&D or QC). The primary goal is documenting measurements in conformity with IEC 60286 standards, in order to simplify the process.

Another welcome and important update to the Klippel Analyzer System is the addition of TEDS calibration data for any Klippel KA3 system that has QC-Card(s) installed. Now, any data stored in TEDS compatible microphones can be easily converted into Klippel sensor files. This information can be used with any input, even those that do not support TEDS, such as the Klippel Analyzer (KA3) Laser card, Distortion Analyzer (DA2), and Production Analyzer (PA) hardware, as well as third-party audio interfaces or sound cards.

For all test scenarios where the Klippel Analyzer Hardware is not required or cannot be applied, Klippel created the QC Stand-Alone Software, which has also now been updated. Used together with the External Synchronization (SYN) add-on, the software is highly suitable for any closed or open loop test scenarios using third-party hardware (e.g., audio interfaces) or wave-file-based analysis. It only requires a PC, laptop, or tablet running Windows and a Klippel USB license dongle.



Fast and cost-efficient quality check of rental Sound Reinforcement Equipment using streaming V/I amplifiers and QC Stand-Alone Software.

The updated QC Stand-Alone Software also supports all tasks and features of the QC software that are based on voltage and current measurements, such as Impedance (IMP) for Thiele-Small (T-S) parameter testing and Motor + Suspension Check (MSC) for patented nonlinear parameter testing (e.g., voice coil offset). This enables users to create powerful and yet price-efficient test setups using smart digital amplifiers with built-in voltage and current sensing, such as the Powersoft Mezzo series, which are capable of streaming sensor signals via Dante network interface for professional quality control in cost-sensitive applications.

Finally, the updated Klippel QC software now provides channel aggregation for multiple microphone measurements. A typical use case for aggregating data is testing microphone arrays, and measuring spatially averaged sound field characteristics.

Aggregation of multi-channel microphone data using a microphone array is a measurement technique often used to acquire data inside vehicle cabins, allowing to assess the soundfield for each passenger. Related to this feature, there is current activity by the AES Technical Committee on Automotive Audio, as discussed at the recent AES Automotive Audio conference that took place in June, in Detroit, MI. The white paper "In-car Acoustic Measurements" suggests a standardized assessment and comparison of acoustic properties in cars. Klippel is contributing to these activities and supports the proposed measurements.

www.klippel.de

NTi Audio

New XL3 Acoustic Analyzer and Sound Level Meter

NTi Audio announced its latest generation handheld measurement solution, the XL3 Acoustic Analyzer. This is a completely new design, evolving from the extremely popular XL2, adding USB-C, wireless web connectivity, a large color touchscreen, and multi-language interface.

The new XL3 Acoustic Analyzer is a Class 1 sound level meter that can serve as a detailed spectrum analyzer, reverberation time analyzer, or sound insulation analyzer, working as a fully contained handheld instrument or remotely controlled and monitored.

The intuitive XL3 user interface is optimized for basic noise monitoring applications as well as offering extensive analysis tools for professionals. The sound level meter is fully networked and therefore allows operation and data access from any mobile device, with 8 hours of battery operation when working in remote wireless mode.

The large 4.3" IPS color touch display offers a resolution of 480 × 800 pixels. This allows the intuitive control of all measurement functions and presents noise level readings in large, clear digits. Users can even select between a traditional white, blue, or black background theme and an additional

haptic keypad with eight buttons controls the common instrument functions, such as power on/off for the XL3 and start/stop for the measurement.

The XL3 includes an integrated web server and file server accessible through the instrument's built-in Wi-Fi or LAN connectivity. This allows users to use any web browser on a smartphone, tablet, or laptop to connect to the XL3 sound level meter through the NTi Connect service. This way, the instrument can be controlled by an authorized user from anywhere in the world. Alternatively, users can connect directly to the device's IP Address through a local network. As an example, this enables to measure sound levels directly from the sending room in building acoustics.

The XL3 forms a Class 1 sound level meter in combination with NTi's measurement microphones M2230 or M2340. All sound pressure levels are measured precisely at the same time and presented in large digits on the color display. Readings exceeding the preset limit turn red indicating non-compliance, while the data is logged in 1 second intervals and the audio is recorded simultaneously. This makes the XL3 a perfect tool for an easy investigation of noise complaints and long-term noise monitoring.

Additionally, the noise spectrum is measured in both octave or one-third octave resolution (e.g., for tonality investigations), as well as the optional event-marker, back-erase and pre-trigger—all of which extend the sound level meter capabilities for effective attended environmental noise surveys.

The XL3 determines the frequency spectrum with a Class 1 filter bank in octave or one-third octave resolution between 6.3Hz and 20kHz. The minimum, maximum, and average sound pressure levels of all frequency bands are measured in parallel. The frequency axis can be freely zoomed and scrolled to 125Hz to 8kHz for appliance noise investigations, for example. The summed level of the selected frequency range is shown in the spectrum view at the same time.

The handheld instrument also measures the reverberation time in octave bands from 63Hz to 8kHz using the standardized T20 or T30. The room may be excited using an impulse source or an interrupted pink noise. The Acoustic Analyzer averages all the recorded decays for each measurement position automatically. Additionally, installing the Extended Room Acoustics Option enables reverberation time measurement in one-third octave resolution, visualizing the decay curve, calculating the spatial average of the room and simultaneous T30/T20/T15/EDT measurement on the XL3 Acoustic Analyzer. Additional quality indicators, such as linearity and curvature, describe the decay curve in accordance with the standard ISO 3382-2. An audio file of the level decay may be recorded for more detailed analysis.

The optional Building Acoustics module enables direct determination of airborne, impact, and facade sound insulation on the XL3. Simply measure in the sending and receiving room and see the sound insulation values displayed for immediate compliance confirmation onsite. For additional partitions with the same source or receiving rooms, the XL3 offers time-saving features by re-using measurement data from previous measurements (e.g., reverberation time data or background noise measurements) for a common receive room. In order to prevent any sound absorption from the user's body, the XL3 can be controlled from a mobile phone using Wi-Fi or the 4G/5G network. Generating the final measurement report for sign-off is easy utilizing the optional Sound Insulation Reporter PC Software.

The NTi XL3 is a complete professional sound level meter and acoustic analyzer dedicated to noise measurements, room acoustics, and building acoustics applications, complemented by a range of optional software modules and functions.

The available optional software includes the Extended Noise Measurement Option for the more challenging tasks of noise consultants; the Extended Room Acoustics Option for extended reverberation time measurements; Sound Insulation Option for building acoustic applications; API Interface Option for integration of the XL3 Sound Level Meter into customer-specific monitoring solutions; and Type Approval Option (pending) for certified measurements.

NTi Audio also offers the NTi Connect 365 free service, which enables the online communication with the XL3 Acoustic Analyzer from anywhere around the world for easy and secure access to webpage and data files, up to 2 GB/month. The XL3 also works with NTi's existing Data Explorer, Room Acoustics Reporter, and Sound Insulation Reporter, PC-based software applications.

As NTi Audio confirms, the new XL3 sound level meter builds on the latest processor and display technologies to provide a comfortable user experience. High-performance hardware packed in a robust housing offers precise 96kHz data sampling in 32-bit resolution. With a frequency range from 0.3Hz to 45kHz, the XL3 is prepared for infra and ultra sound measurements. With single-range measurement and the automated sensor detection reading the electronic data sheet from the connected measurement microphone, the XL3 Acoustic Analyzer is indeed the equivalent of a Swiss Army knife for noise measurement professionals.

www.nti-audio.com

Echo Test + Measurement

New AIO Modules and Software Version 4

Echo Test + Measurement released Version 4 of the Echo AIO Test System Software now adding support for additional sample rates, a revised user interface with support for all the new AIO features, automatic firmware updating, and Windows 11 support. Previously the AIO only supported 48kHz, 96kHz, and 192kHz; the new firmware now also supports 44.1kHz, 88.2kHz, and 176.4kHz. The AIO device configuration is now automatically and persistently stored in onboard flash memory, and the update adds support for the H-Module and C-Module. A total of six module types are now fully supported in the Control Panel, CLI, and API and users can combine any two analog modules (A, S, L, or H), or combine any single analog module with either a T-module or C-module.

Other updated software features include the possibility to select RMS and peak hold metering ballistics, set peak output voltage limits on each analog output, and

A TEDS reader on all analog inputs allows copy and paste of TEDS parameter values to other applications. The CLI and API library also support exporting TEDS data to JSON and CSV formats.

The T-module supports I²S and four-channel TDM, the firmware no longer needs to be loaded manually, and the logic level is now software selectable (1.8 V or 3.3 V).

The AIO intentionally no longer supports USB audio volume controls, ensuring that the muting or lowering the system volume level does not affect the output level of the AIO.

The Control Panel, CLI, and API library are all available for both Mac and Windows systems. A single comprehensive installer for either Mac or Windows simplifies loading or updating the software. The Echo AIO software package includes the AIO Windows driver (USB 2.0 Audio Class) with supports for ASIO, WASAPI, DirectSound, and Wave APIs, enabling virtually any audio test application to interface with the AIO. No driver is required for Mac OS systems.

Among the latest modules, Echo Test + Measurement announced the release of the AIO C-module designed to interface with common test fixtures used in testing hearing aids, hearing assistance devices and other wearables. The AIO C-module features Advanced Battery Simulation and Monitoring (ABSM) technology that provides an adjustable 5V @ 1A DC power supply for simulating battery under-voltage or over-voltage conditions and a current monitor that covers a range from 1µA to 1.25A with zero offset. The C-Module also has eight GPIO inputs and eight GPIO outputs for controlling isolation chambers, pass/fail indicators, and other equipment, a fixed 5V @ 1A DC power supply for powering test fixtures and a pressure, temperature, and humidity sensor for establishing environmental baselines in a production environment.

Echo Test + Measurement also redesigned its AIO L-module, featuring four IEPE mic/line inputs and two differentially balanced line outputs for general purpose electronic and acoustic testing that requires more than two input channels. The AIO L-module is available in single module (AIO-L1) or two module (AIO-L2) configurations or can be mixed with other AIO modules to create a customized solution. Each input can be individually configured as a line level input or as a measurement microphone input. Microphone inputs include an IEPE microphone conditioning amplifier with integrated constant current power supply (CCP)

The original L-module output circuit was designed as a hybrid line output and headphone driver (for small headphones with minimum 32Ω impedance). With the introduction of the new H-Module designed specifically for headphone testing, Echo redesigned the L-Module to be a true line output stage optimized for minimum 600Ω loads. The new line output circuit has significantly better noise, distortion, and dynamic range specifications.

https://echotm.com

When coupled with an Echo AIO audio test module (A, L, S, or H), the new C-Module offers a powerful measurement and control system for wearable devices, hearing aids, or hearing assistance products, in one convenient unit.

The new Echo AIO H-module now offers a two-channel acoustic interface designed for production testing of wired headphones and earbuds and verification of sub-assemblies. It features two IEPE mic/line inputs and two single-ended headphone outputs with integrated impedance measurement.

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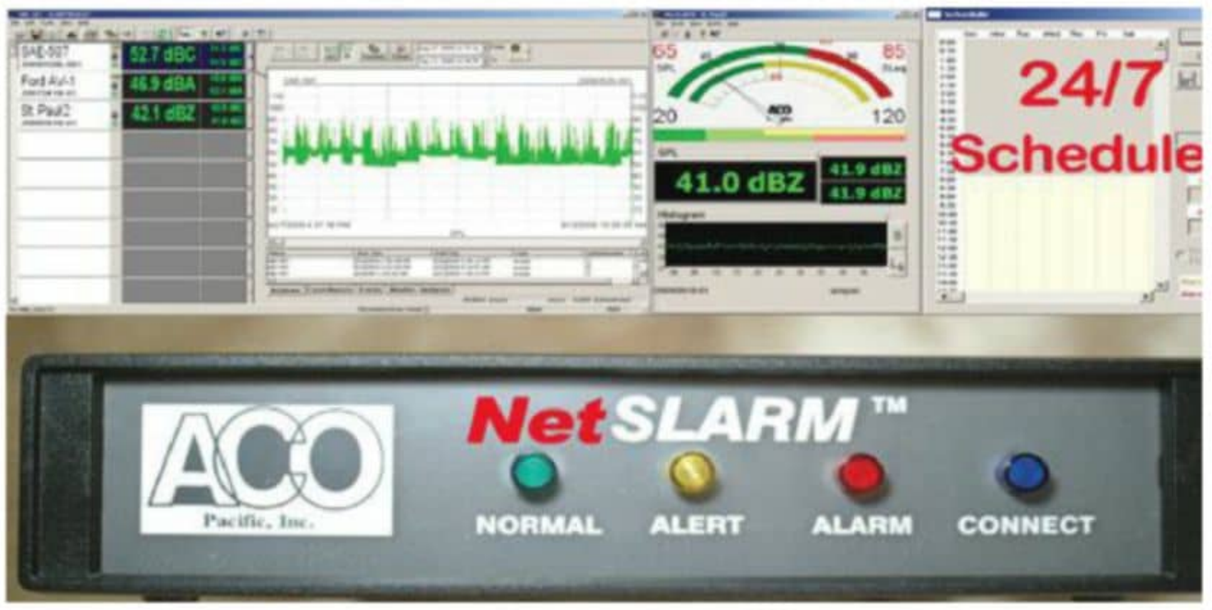

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Acoustical Measurement and DSP Application in Automotive Audio

This article offers a primer on how to approach acoustic measurements and the practical application of DSP in car cabins, one of the hardest acoustic environments for quality sound reproduction. The topics addressed in this article will be expanded in future articles.

By
Philipp Paul Klose

Car cabins are one of the hardest acoustic environments to create quality sound reproduction in: The space is small and has massive room resonances from the deepest bass notes up until the midrange frequency band. Acoustic engineers are confronted with surfaces that either create nasty reflections (e.g., glass or metal), while other parts are covered with effective acoustic dampening materials (e.g., Thinsulate or Basotect), and are therefore very absorptive. At the joints of interconnected materials rattle noise can occur.

The speakers can only in a few cases be directed at the listeners at the 0° axis and the car is producing noise by itself through flowing wind, rumbling tires, and a working powertrain—be it electric or of an internal combustion type. It seems like the only good thing is that we know where the listeners ears are positioned and that they will barely move their heads.

Although it is possible to have good sound reproduction in the car: It is just a little bit harder to attain than in other environments. Thankfully, DSP power is available, and it is more than useful in cars. While tuning often seems like a black art and is feared by many engineers, much like an exam in higher mathematics, it is not that complicated.

Proven measurement techniques help substantially with achieving a good tuned system and mastering the many tasks that can be also done by engineers who are not blessed with "golden ears."

The goal of this article series is to give you a primer on how to approach acoustic measurements and the practical application of DSP in an automotive environment. Music is the most important and prevalent source used for tuning (plus it is free of regularities). That is why it will serve as lead source here. Other audio sources such as warning tones, engine enhancement sounds, and voice assists are calibrated with the same approaches and techniques, but pose challenges in other areas and are sometimes loaded with boundary conditions through external regularities.

Setup

To start the calibration, a properly prepared car is the groundwork. The loudspeakers must work, be mounted in the right place, and the playback chain must work. The integration of speakers (be it in DIY or mass market) in the car and the implementation of automotive infotainment architectures and transmission protocols such as MOST, A^{2}B, and CAN is a topic on its own—we will take that as given here. A word of safety before we start: Never measure, tune, and drive at the same time!

Microphone Techniques and Their Caveats

Every existing automotive microphone technique does have its pros and cons. During calibration and decision making, it is important to know the potential and possible limitations of your technique.

The boundary conditions reflect on the choice here: Depending on how much time you want to spend on measurements, the calibration source, your focus, desire for accuracy and precision, or if you want to or can sit in the car during calibration affects on your choice here.

Variant A: Single Microphone Fixed in Place

This is the simplest measurement technique available: The microphone is put into one place with a short microphone stand and stays there during the duration of the measurement.

During the tuning process, data from this single point can be used for sensitivity checks of the loudspeakers (how much voltage leads to which level of SPL), polarity checks of the playback chain, and log sweep analysis.

Note that log sweeps analyzed with discrete Fourier transform (DFT) from fixed single point microphones are per se correct in its levels, but the acoustic resolution in the frequency domain is limited. Especially in near fields as you find them in car cabins, levels change dramatically with small positional changes.

Due to the 20dBr errors from only a single data point, results for to-be-made tuning decisions can be misleading and lack reproducibility. Errors occur mostly in the mid and high frequencies and then ultimately lead to mediocre DSP settings and sound in your car.

The missing resolution in the frequency domain does not affect other drawn conclusions from log sweeps such as group delay, THD+N, waterfall, RT60, and more.

What speaks for the "single microphone fixed in place" method, is the fact that you do not have to be in the sound field yourself. Think here of very loud sound systems and long sessions. For an improved resolution in the frequency domain the moving microphone method is a good choice.

Variant B: Moving Microphone (with Noise for RTA Frequency Response)

For this moving microphone technique (sometimes also called a "flaming torch"), the microphone is moved slowly around the listening space for 10 to 30 seconds. The stimulus used is a noise with a fast Fourier transform (FFT) real-time analyzer (RTA) used for analysis. For an example, REW's free RTA is very capable of performing this task either with signals directly fed into the application via an audio interface or with pre-recorded files drag and dropped into the RTA window.

The result of the moving microphone method is in the frequency domain more accurate when repeated then the fixed in place single measurement technique (several measurements of the same acoustic event will give you same results). A negative aspect of this method is that the measurement requires you to be in the car, which can be problematic with very loud and long measurement sessions. Also, you could alter the measurement through your breathing, wind noise (that is created through moving the microphone), and cable noise. This must be kept in mind as potential problem.

As a starting point for the FFT, use 1/24 smoothing, 512k FFT length, no averaging, Hann windown and 50% overlap with pink noise as stimulus. It takes approximately 11 seconds to finish the measurement with these settings. Within this time, move the microphone around your head where your listening positions are [1].

What Is the Resolution of My Microphone Array?

The filter reconstruction measurement is used to verify the resolution of an acoustic measurement chain.

Baseline measurement - measurement with known change in frequency domain = reconstructed resolution of applied filter (blue).

The applied filter electric signal is additionally displayed (green) in **Figure 1** as the target. The further the reconstructed filter is "away" from the applied filter, the less resolution your measurement chain has.

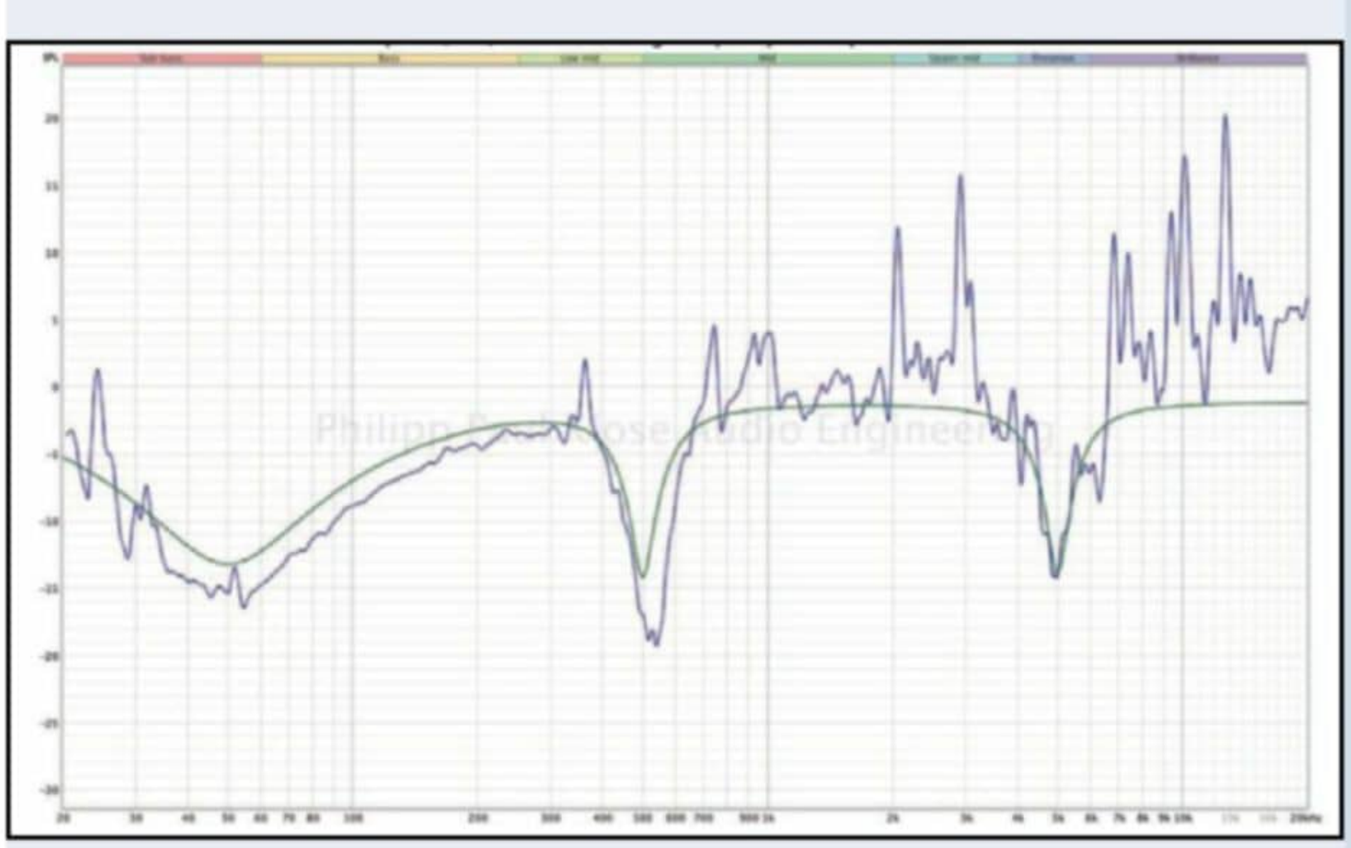

Figure 1: Filter reconstruction measurement. In the acoustical sound field, the applied filter (green) vs. the reconstructed filter from one microphone data point (blue). Errors in the high-frequency band up to 20dBr.

Automotive Sound

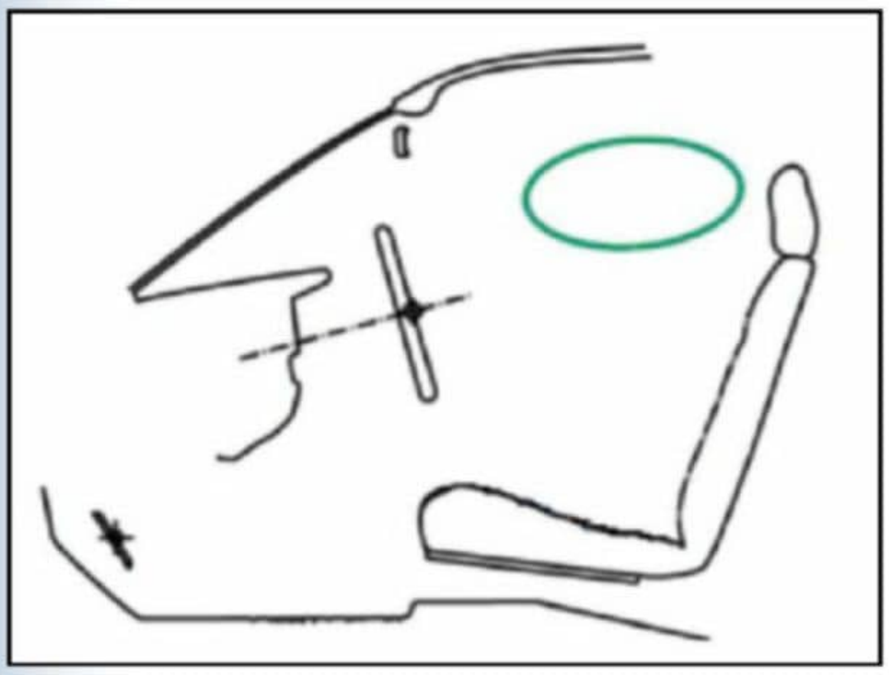

Figure 2: Geddes discovered that six equally spaced microphones placed where the ears of potential customers are (from small to very big persons, in the so-called “ear ellipsoid”) with their data points averaged, gives the best measurement result in the cars in terms of resolution, accuracy, reproducibility, and coherence with personal perception. (Image Source: *Adjustment of Auditory Displays for Driver Assistance Systems* by Strauß, Treichel, Kesseler)

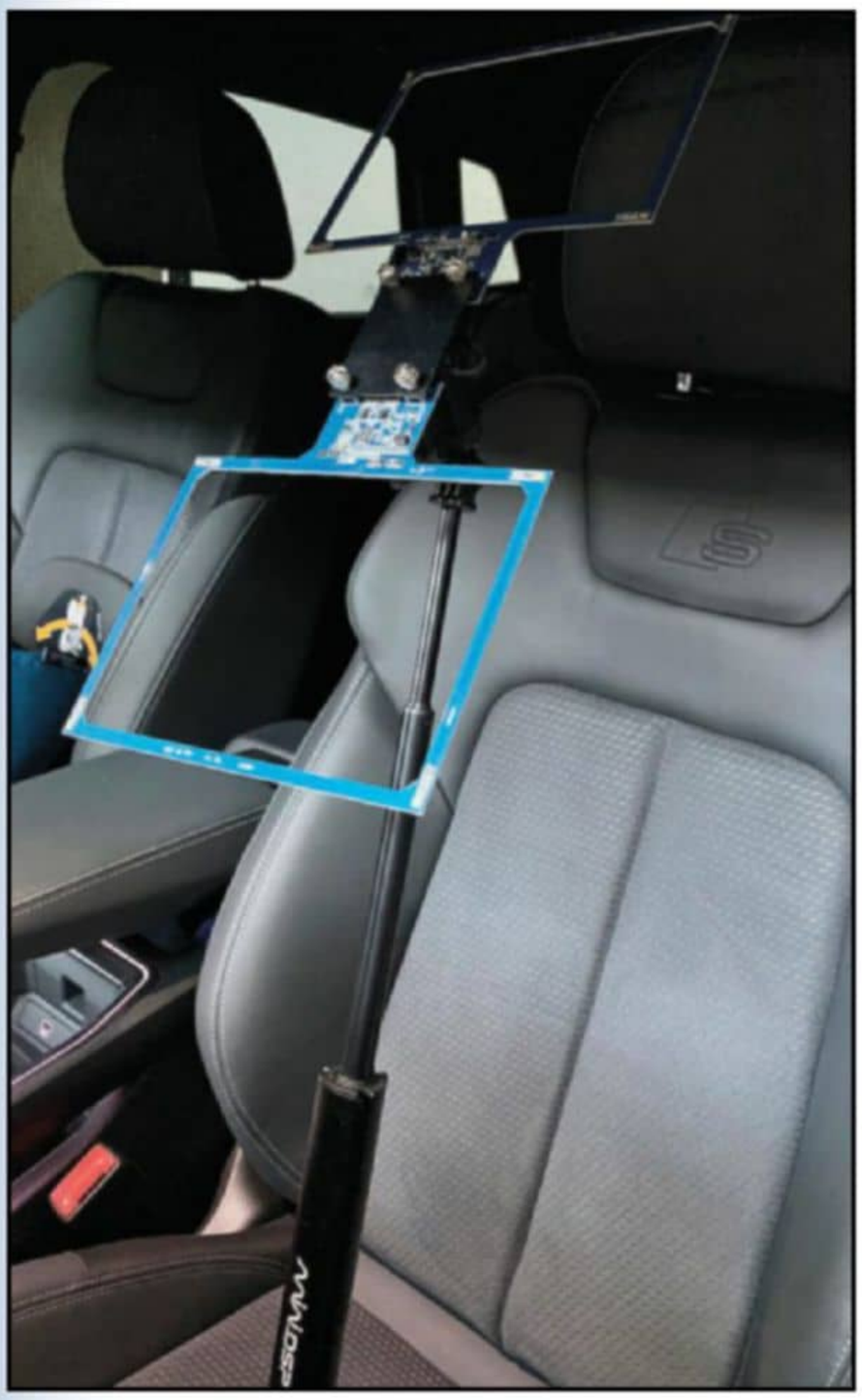

Photo 1: MiniDSP’s Umik-X is made for measurements with car cabins in mind. One group of 2×4 MEMS microphones form an array for one seat, thus increasing the resolution to the original six-point setup proposed by Geddes.

With the “RTA” mode enabled, REW makes a level correct measurement and compensates for FFT gain and the 3 dB/octave loss from the pink noise. So, you can aim for your defined target curve in your tuning and even out troughs and crests in the frequency response. Note that the moving microphone method cannot be used with sweeps! So information such as total harmonic distortion (THD), waterfalls, and group delay cannot be drawn from this method.

Variant C: Array Setup (for Noise and Sweep Stimuli)

The gold standard. Array measurements in cars go back to the work of Earl Geddes who proposed the “localized sound power method” in the early 1980s when he worked for Ford [2]. Geddes discovered that six equally spaced microphones placed where the ears of potential customers are (from small to very big persons, in the so-called “ear ellipsoid” in green, see **Figure 2**) with their data points averaged, gives the best measurement result in the cars in terms of resolution, accuracy, reproducibility, and coherence with

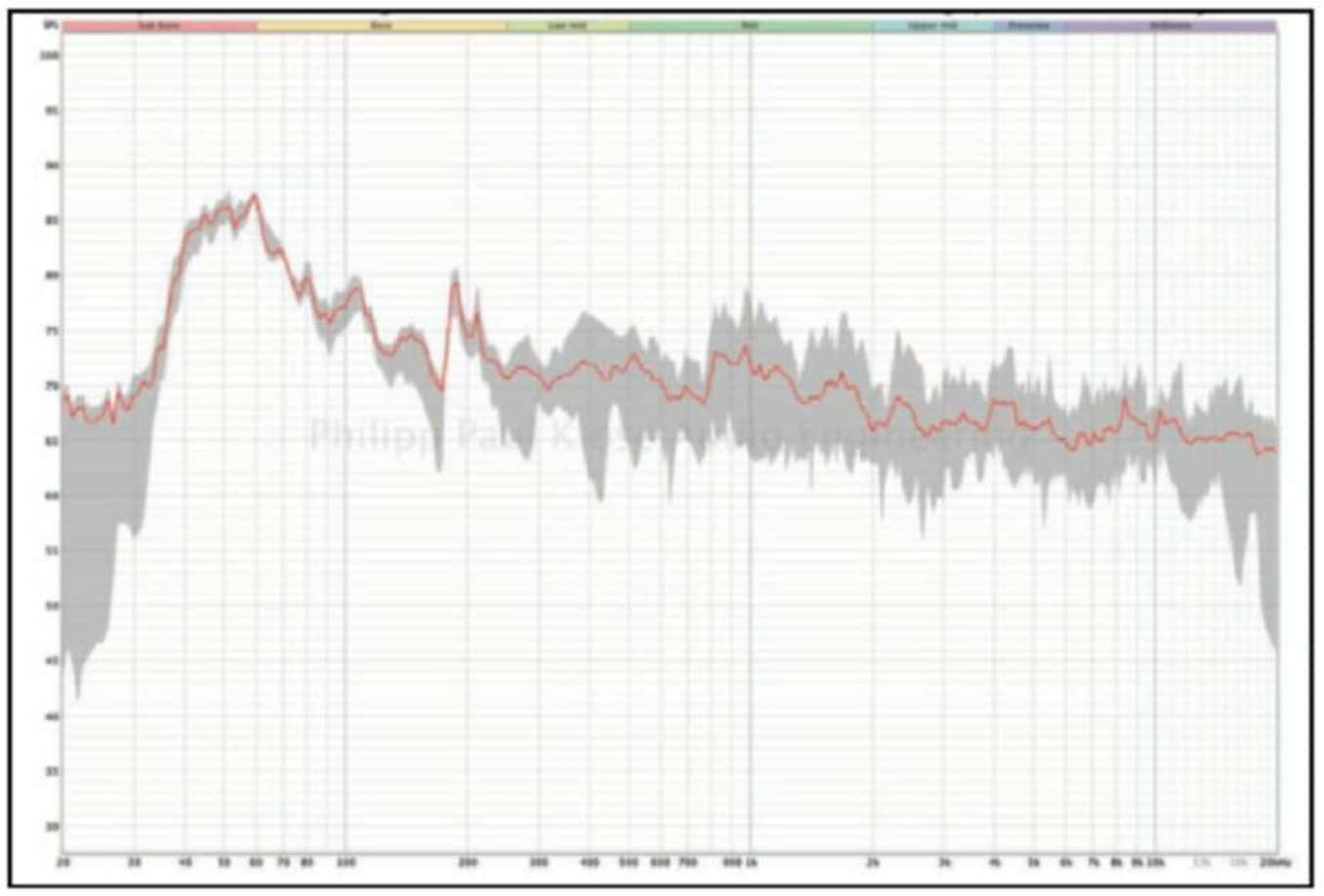

Figure 3: An array measurement. The average of all measurements (red) and minimum/maximum variations are shown as a grey tube around the average.

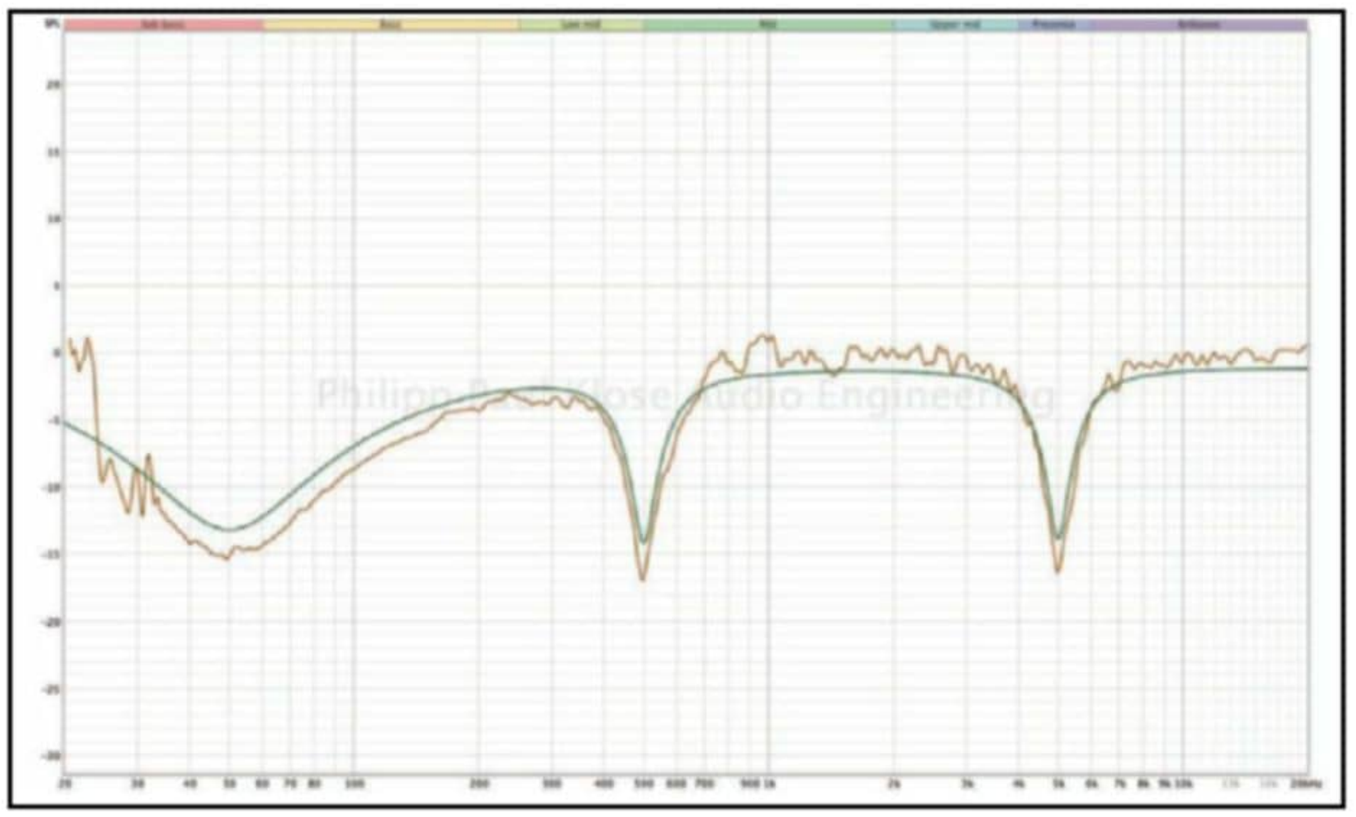

Figure 4: Filter reconstruction measurement. In the acoustical sound field, the applied filter (green) is shown vs. the reconstructed filter from the Umik-X array. The error is 2dBr.

personal perception. This measurement technique is often used at car manufacturers and car audio brand suppliers. It is said that the six-microphone array gives a reliable resolution of 1/3 of an octave equivalent [3].

As an out-of-the-box solution, MiniDSP's Umik-X can be named here (**Photo 1**). It is made for measurements with car cabins in mind. One group of 2×4 MEMS microphones form an array for one seat, thus increasing the resolution to the original six-point setup proposed by Geddes.

Working with arrays gives the greatest confidence and reproducibility in the tuning process. They can be used with log sweeps and noise. Array results are acquired faster and are easier to read, and easier to work with than single microphone or binaural head results, especially for beginners and with REW as an analysis tool (**Figure 3**).

Since arrays are taking the seat of listener, you do not have to be in the sound field yourself. Note that when you are using an array, only one microphone signal of the array must be used for delay measurements! With arrays, a high acoustical resolution can be achieved (**Figure 4**).

Variant D: Binaural Head Setup (for Noise and Sweep Stimuli)

Automotive acoustic engineers are not only working on sounds that are created through

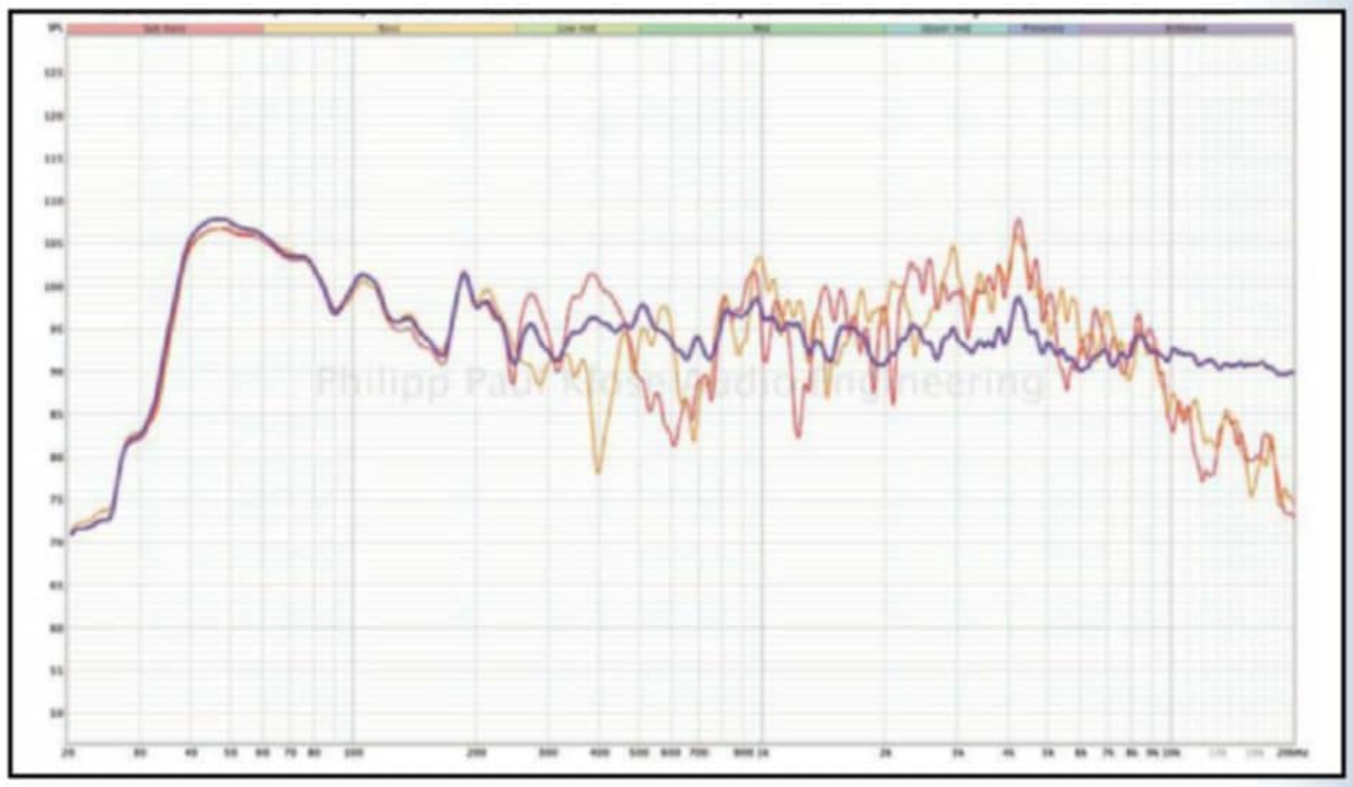

Figure 5: The array measurement (violet) vs. the artificial head measurement (orange and red). While the congruence of the array and the artificial head measurement is high up to the Schroeder frequency of 150Hz , above 150Hz the spatial instability (e.g., at 400Hz) and the HRTF of the artificial head (f<2kHz) make the measurement results of this system measurement of a production car hard to read.

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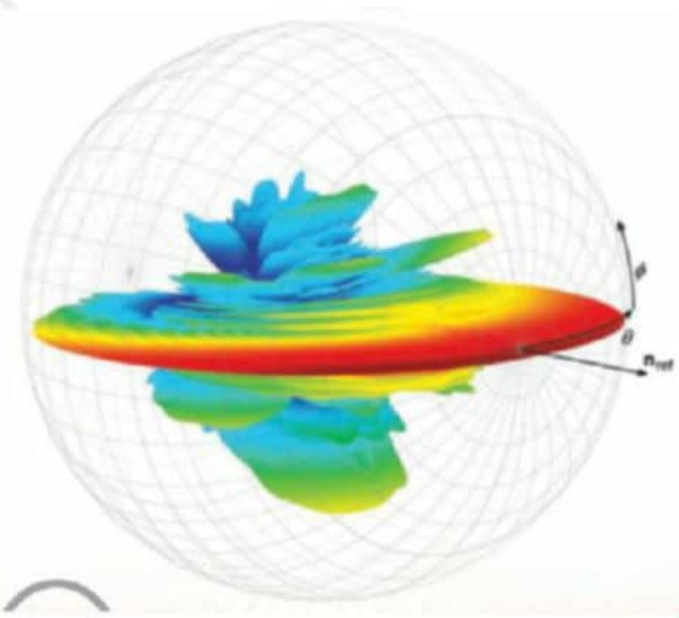

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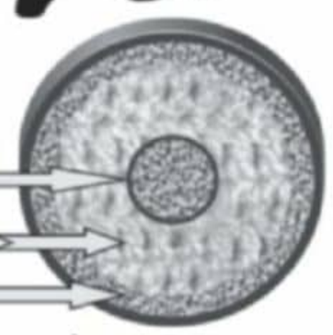

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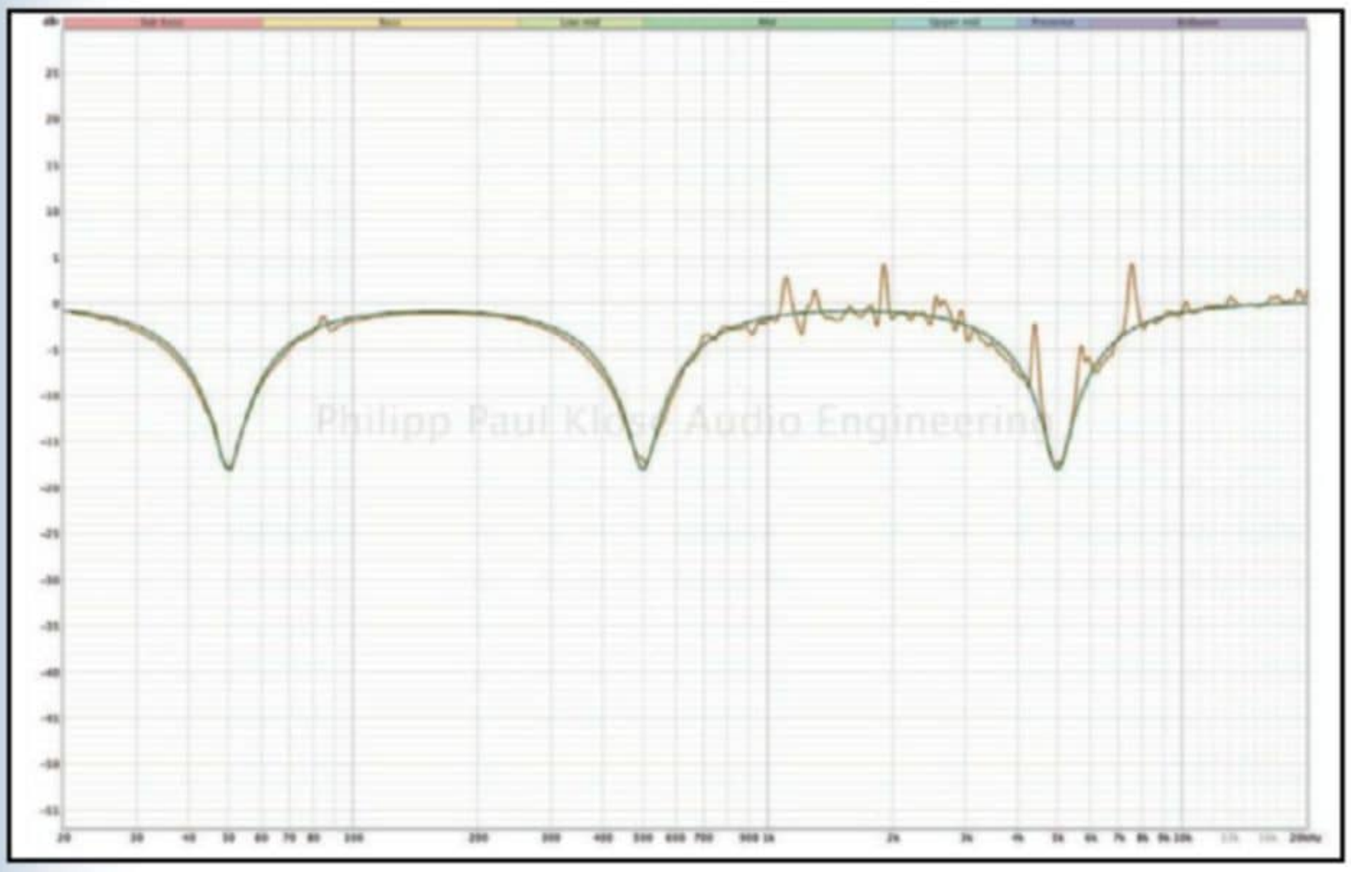


Figure 6: Filter reconstruction measurement with the average signal of a binaural head (brown) and imposed filter on the spectrum (green). The relative error is up to 4.2dBr.

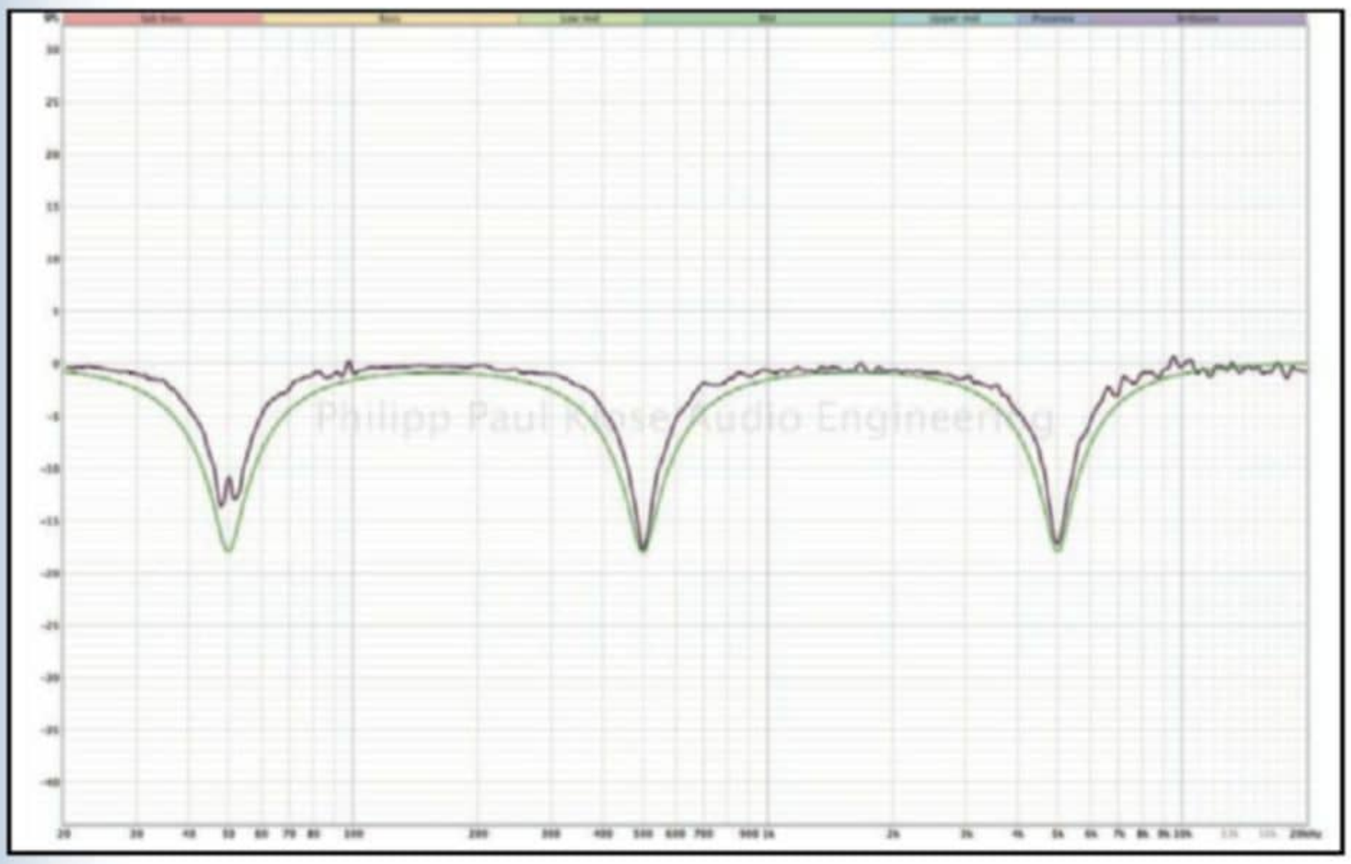


Figure 7: Filter reconstruction measurement of a Sennheiser Ambeo headset (dark brown) and imposed filter on the spectrum (green). The relative error is up to 5dBr.

References

[1] Real-world application of the moving mic technique can be found at: www.youtube.com/watch?v=6Riuwqzjql Q

[2] E. Geddes and H. Blind, "The Localized Sound Power Method," Audio Engineering Society, Paper No. 2127, October 1984 www.aes.org/e-lib/browse.cfm?elib=11627

[3] M. Strauβ, T. Treichel, and R. Kessler, "Adjustment of Auditory Displays for Driver Assistance Systems https://pub.dega-akustik.de/DAGA_2010/data/articles/000235.pdf

[4] "OpenAural," MakerBot Thingiverse, December 2020 www.thingiverse.com/thing:4691843

speakers, but also sound events that are determined by physical properties of used parts or are unwanted in general. These are, for example, intake noise, engine vibrations, or interior trim pieces that rattle. These sound events (e.g., noise, vibration, and harshness or NVH) are measured and influenced by NVH engineers. Traditionally, an important tool for NVH engineers are binaural heads. A binaural head simulates a human head, torso, pinna, and ear canals.

For sound system measurements, the results of these heads are from an acoustic resolution standpoint better than the single microphone fixed in place approach, since you have two microphone signals and a (well-defined) head-related transfer function (HRTF) through delay (distance between the microphones) and acoustic shadowing (through the form of the torso) added (**Figure 5**). For decision making in electro-acoustic applications, HRTF and the somewhat limited resolution of the binaural head, especially in the mid and high frequency, make binaural head recordings hard for beginners since a suitable correction function must be applied to the measurement. (**Figure 6**).

For an accurate measurement in a car, several head positions must be averaged. Signals from binaural heads can be used for later reference and storing of sound impressions through impulse responses, noise and reference track recording.

Variant E: Yourself as a Binaural Head (for Noise and Sweep Stimuli)

When your budget is too constrained for a professional-looking fully fledged binaural head or a microphone array, you can use yourself as a binaural head.

As with torso-based binaural head recordings, the same principles apply here: The number of data points is relatively small and a HRTF is added. But, in contrast to torso binaural head recordings, you can move and point and reach different positions quicker and easier. I am using a pattern following this track: looking straight ahead, left mirror, right mirror, sun visor, and bottom of steering wheel for a few seconds.

This is an interesting method when you want to travel with light equipment or want to use as little equipment as possible and can be in the sound field during the measurement yourself (**Figure 7**). When used with noise and when you are moving your head during the measurement, this method provides a lot of data points with noise and sweeps as stimuli, in a short amount of time. It is also advantageous to use the binaural head method for future reference

Photo 2: A good working combination is a HEAD acoustics artificial recording head combined with a Umik-X array with two miniDSP Uma-4 modules.

and storing of sound impressions through impulse responses.

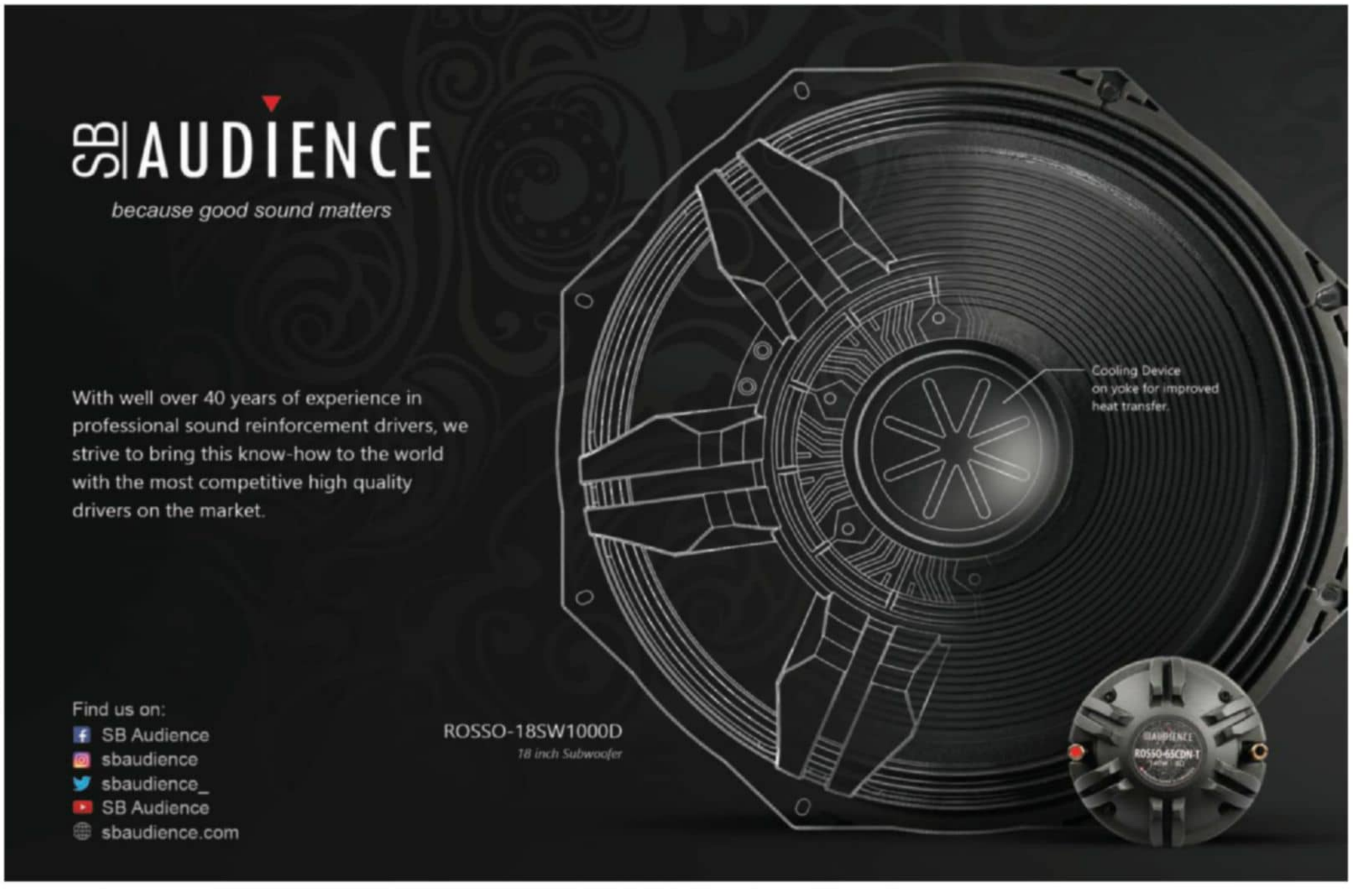
Variant F: Mixed Setup (for Noise and Sweep Stimuli)

There is nothing wrong with combining these approaches. A combination of methods can make sense, depending on your special requirements or goals.

A good working combination is a HEAD acoustics artificial recording head combined with a Umik-X array with two miniDSP Uma-4 modules (**Photo 2**). The signals of the artificial head and the Umik-X are matched through calibration files, averaged, and analyzed together [4].

About the Author

Philipp Paul Klose is a senior engineer for automotive audio concepts and functions and previously served in positions such as audio system development engineer, project lead, and infotainment test engineer. He holds a Bachelor of Arts (Hons.) degree in Audio Production. Within his 10 years of automotive experience, he has worked for companies such as Audi, Volkswagen, Porsche, Lamborghini, and Ferrari.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Practical Test & Measurement

Transitioning Audio Tests from R&D to the Production Line

Production line audio testing poses many challenges such as noisy environments, harsh operating conditions, high throughput, relative limits, and more. In this article, Steve Temme shares his observations and outlines the main considerations to ensure a successful operation.

By
Steve Temme
(Listen, Inc.)

Your product design is finalized, your prototypes are close to perfect, and your audio product is ready to move to production. But how do you ensure that the product hitting the retail shelves sounds exactly as you intended? Your production line is probably thousands of miles away, with time zone and language complications. Pandemic-related restrictions make it hard to visit to configure, supervise, and train factory personnel. Furthermore, production is often outsourced to a contract manufacturer, adding another layer of complexity to both culture and information transfer.

Every production facility has one priority: to build products that meet the specifications with the greatest efficiency and highest yield possible for optimum profitability. Think about what that means for end-of-line tests. Their goal is to have as many units pass as possible. Tests must be fast to avoid slowing down the production line, and the test environment is generally noisy, often with variable temperature and humidity. The quest for high yields and manufacturing efficiency, along with challenging operating conditions and sometimes poor training, can result in unreliable measurements. When this happens, the only loser is you and your brand's reputation.

I'll never forget being told about a production line operator who increased pass-rate by opening the test box during measurements so that the factory background noise would mask the high-order harmonics to fool the perceptual Rub & Buzz algorithm. This might improve his yield, but it's not so good when these products ship to customers! While this is one of the more outrageous violations of end-of-line test protocol, I have also seen many other questionable measurement practices over the years. In this article, I'll try to de-mystify production tests and look at some simple measures that you can take to ensure a smooth transition.

Production Line Test Considerations

Production line measurements differ considerably from R&D testing. In the design phase, you carefully select transducers and components, engineer signal processing algorithms, measure performance with different materials and components, and measure and re-measure parameters that affect the listener experience such as frequency response, directionality, distortion, and more. These measurements should be calibrated, and are often made in an anechoic chamber or with a test system that replicates anechoic conditions using a simulated free-field technique. The environment is usually fairly quiet, and can be controlled when making measurements (e.g., by turning off noisy air conditioning or refraining from talking). The time taken to make and analyze measurements is usually not critical, and you can use time-consuming workarounds, such as exporting data for additional analysis to extend the capabilities of your test system.

There are several major changes when you move into production. Speed is critically important, whether your measurements are in-line on the conveyor belt, or offline in a test box. Factories are inherently noisy and both the physical setup and test system need to minimize the effect of background noise. Any product is likely to measure differently under production line conditions compared to the laboratory. For this reason, end-of-line measurements are generally relative to either a golden unit or statistical average rather than absolute measurements that can be reproduced in any test lab. Limit setting is tricky and needs to be approached carefully, taking into account a variety of factors. Finally, different parameters such as Rub & Buzz and loose particles are measured to account for manufacturing-induced defects.

Speed vs. Accuracy

Most transducer production line tests are completed in a second or two. While there are ways to optimize tests for speed, the biggest factor is probably the test system. Speed is one of the main differentiating factors between lower cost "hobbyist" test systems and a true professional-grade product. While most manufacturers claim "fast testing," it's important to ascertain how long it will take to make the specific measurements required. For example, can the system make both perceptual and traditional measurements simultaneously with just one sweep of the stimulus, or are two test sweeps needed, doubling the test time? Can electrical measurements such as impedance be measured at the same time as acoustic properties? Are the algorithms optimized for fast measurements? Will the results be accurate and repeatable with fast sine sweeps, or is it necessary to sweep the signal slower to get consistent results? You can create a test signal as short and fast as you like, but if the test system cannot return repeatable results at that speed, it's pointless!

Aside from the test system, there are other techniques you can use to minimize test time. These include sweeping from high to low frequencies to minimize transducer settling time, sweeping only over the frequency range of interest, and using the lowest resolution necessary to obtain repeatable and reliable results, which may vary across different parts of the frequency range.

Noise Immunity

Noise is rarely a problem in a laboratory setting. Measurements are often made in an anechoic chamber or a test box, and even in an untreated room, the environment can be controlled. This is not the case for production lines. Background noise in the region of 80dBSPL to 100dBSPL is common, especially at low frequencies and there is usually a mixture of continuous noise (e.g., machinery hum or air conditioning) and transient noise (e.g., air compressors, intermittent machinery noise, etc.). These background noises can mask production line defects—as my manufacturing friend once demonstrated, it's easy to get a speaker with bad Rub & Buzz to pass if you open the test box and add a little background noise to mask the high-order distortion harmonics.

A significant factor in mitigating background noise is the test system itself. Good production line test systems feature filtering algorithms designed for noise immunity as well as speed. This has always been a focus at Listen, with the introduction of many unique algorithms to minimize the effects of background noise throughout our entire 27 year history. This is illustrated in **Figure 1**, which shows the repeatability of SoundCheck's enhanced Perceptual Rub & Buzz (ePRB) algorithm, specifically designed for production line tests, vs. a perceptual Rub & Buzz algorithm from a different

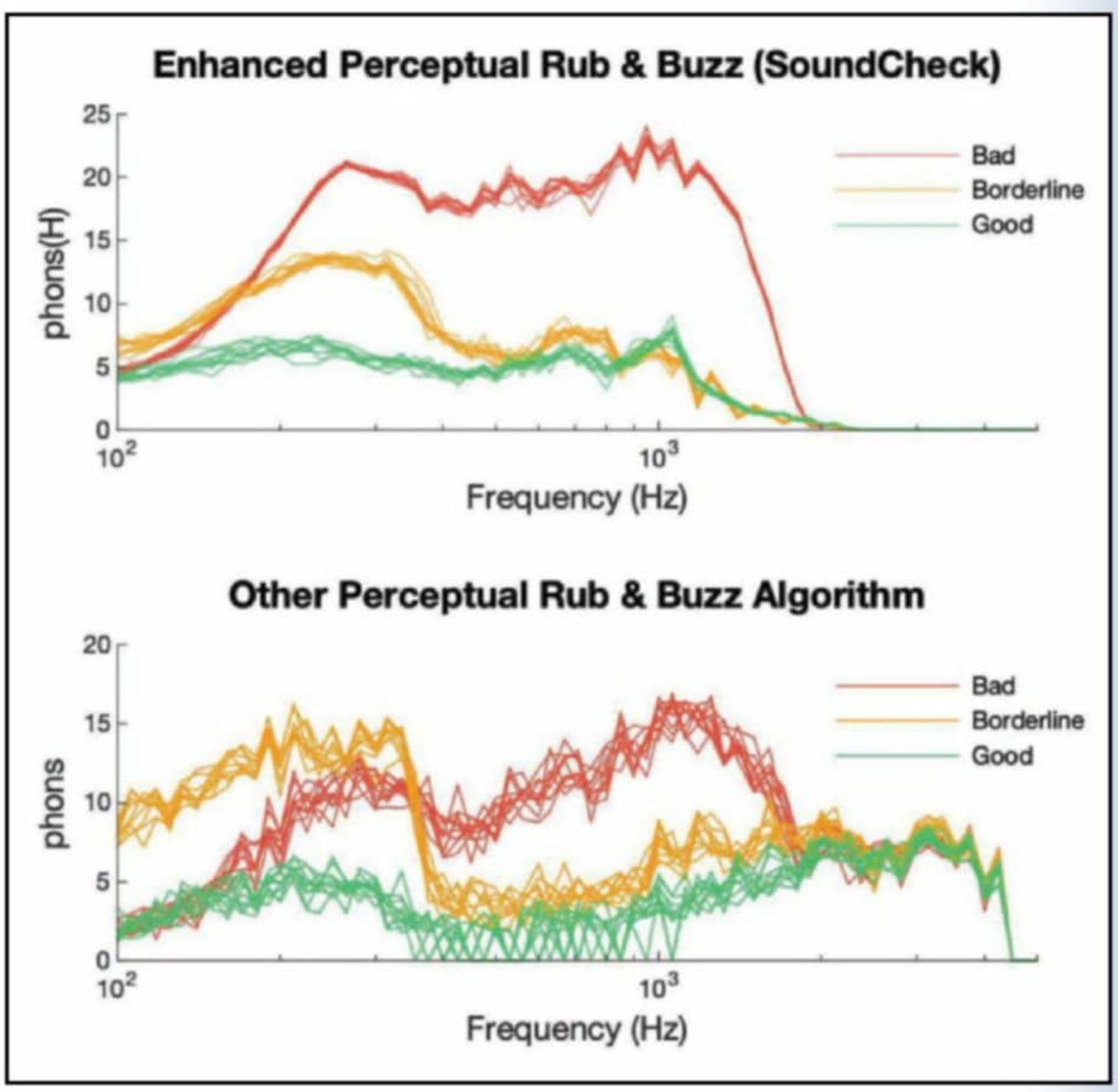


Figure 1: Three speakers were each measured 10 times in an ordinary room using both SoundCheck's enhanced Perceptual Rub & Buzz algorithm with noise reduction and a perceptual Perceptual Rub & Buzz algorithm from a different manufacturer. The improved repeatability with the noise reduction algorithm is clear.

manufacturer. Interestingly, this is where I often see manufacturers who have designed their own test systems stumble—it's much easier to write a Matlab or Python algorithm that measures correctly in a quiet environment than it is to write one that will measure it accurately in a noisy factory.

Acoustic isolation is also significant in minimizing background noise. Simple products such as transducers are usually tested in-line, with acoustic panels around the test station. Products that require more complex fixturing, such as VR headsets, headphones, and so on are usually measured offline in an acoustic test box. These test stations generally offer better acoustic isolation, but correctly designing a test box is no easy task. It is important to minimize reflections, from both the test box and the fixture, particularly those that cause sharp reflections or dips, as these affect measurements the most. This can be particularly challenging when there are metal test fixtures inside the chamber. Higher frequencies also pose reflection challenges; since the wavelengths are small, minor differences in microphone positioning change the frequency of the reflection, making it hard to create limits.

There are many systems integrators specializing in test boxes and in-line test equipment that can usually combine your test system of choice with their hardware. **Photo 1** shows the AB41 end-of-line test chamber from Bojay, a China-headquartered systems integrator that works with many manufacturing facilities worldwide. This particular chamber has internal dimensions of approximately 0.5m × 0.5m × 0.75m, and is designed for testing small products such as drivers, smart devices, cellphones, VR headsets, and so forth. It can attenuate external background noise by up to 40dB inside the chamber, reducing factory noise levels sufficiently to provide accurate results. The internal mounting fixture can be customized depending on the product.

Photo 1: Bojay's AB41 end-of-line test chamber is a fairly typical production test chamber for offline measurement of small audio devices such as drivers, smart devices, cellphones, VR headsets, etc.

Limits

As I mentioned earlier, production line measurements tend to be relative rather than absolute, as differences in test fixtures, background noise and measurement type make it virtually impossible to correlate production results with R&D measurements. Typically, production measurements are made relative to a golden unit or a statistical average, with floating limits. The golden unit may need to be frequently re-measured, and limits re-calculated if environmental conditions such as temperature and humidity vary significantly over time.

Limits may need to float in both level and frequency to overcome reflections from fixturing and from the test box itself. When configuring limits, devices need to be measured multiple times for Gage R&R to understand any variation in the product measurement positioning and to ensure measurement repeatability. There is often a certain amount of trial and error involved in setting limits, tweaking them until there is acceptable yield and good correlation with human listeners. Correctly setting limits is an extremely complex process and to cover it thoroughly would be a whole article in itself. When beginning production of your device, it's well worth being on-site to assist with limit-setting, so that you can truly see and hear what is going on.

There is some hope that limit setting will become simpler in the future with the trend toward artificial intelligence and perceptual models for characteristics such as Rub & Buzz and loose particles. These measurements have simple level limits that usually remain constant across the frequency range, making them extremely fast and simple to configure.

What to Measure

Most laboratory measurements are not relevant on the production line. Conversely, there are measurements for manufacturing-introduced defects that have probably not been measured during product development.

Production line tests, as shown in **Figure 2**, measure overall performance parameters such as frequency response and THD to ensure that it is within specification, and also make measurements

to identify manufacturing errors, such as reversed polarity, Rub & Buzz and loose particles. **Table 1** shows typical manufacturing transducer tests, along with the rationale behind measuring these parameters. This will vary depending on your specific product, but a list of specific measurements is a good starting point as you start to develop your production tests.

While the majority of test methods have remained similar since their introduction in the late 1990s and early 2000s, the industry is shifting toward perceptual measurements for identification of manufacturing defects. Perceptual measurements use a model based on the human ear to identify only those faults, which are audible rather than harmonic or time-based analysis to identify all defects, audible or not.

These methods offer two significant advantages over traditional methods. Most significantly, they result in higher yield with no perceived quality degradation, as products with inaudible defects are not rejected. Perceptual methods are also very simple to configure and set limits. The limit is typically a flat line across the frequency spectrum based on a known level of audibility which often will remain constant across multiple products. That said, there is valuable additional information about your production line to be gleaned from also making conventional distortion measurements. Some test systems can do both simultaneously with no increase in test time, giving you the best of both worlds.

Test Equipment

We have already discussed how production line test equipment needs to be fast, noise-immune, and flexible to offer the range of tests that you need now and in the future as technology evolves. But what are the other distinctions between a production test system and the configuration you might be used to in the laboratory? Generally speaking, production applications require a much-reduced feature set, as flexibility for a wide range of highly customized laboratory tests is not needed. Because of this, the price-point is lower.

While some manufacturers promote a separate low-cost system for end-of-line tests, at Listen we have taken a modular approach to both software and hardware, with a production system coming in at approximately 50% the cost of a laboratory-grade system. This approach maintains seamless compatibility between manufacturing and development, so tests can be created and modified using an R&D system and sent to production for implementation. Conversely, production line results can be emailed back to the R&D lab for more detailed analysis to aid with any troubleshooting. This modular approach also ensures future compatibility. Technology is evolving at a rapid pace, and the ability to adapt to new digital interfaces such as A^{2}B, or increase the channel count of your test software to measure microphone arrays that might be required for a new product, offers peace of mind.

Production lines also have different hardware requirements. In the laboratory, flexibility is key, and engineers are generally fairly comfortable with changing the setup as needed. Once you get into production, ease of use and error-proofing become the driving factors as the test operator likely has minimal acoustic knowledge. Incorrect setup or accidental configuration changes might not even be noticed unless units start failing.

We've promoted software and sound-card-based systems for more than 25 years now, and although the third-party audio interfaces we recommend, such as RME and Lynx, offer excellent performance, they have their disadvantages when used in a production setting. Between the computer, sound card and the ancillary components such as amplifiers, microphones and microphone power supplies, there are many cables that need to be connected correctly. These professional sound cards also have many front panel controls, including gain adjustment. You can probably imagine what happens if a gain

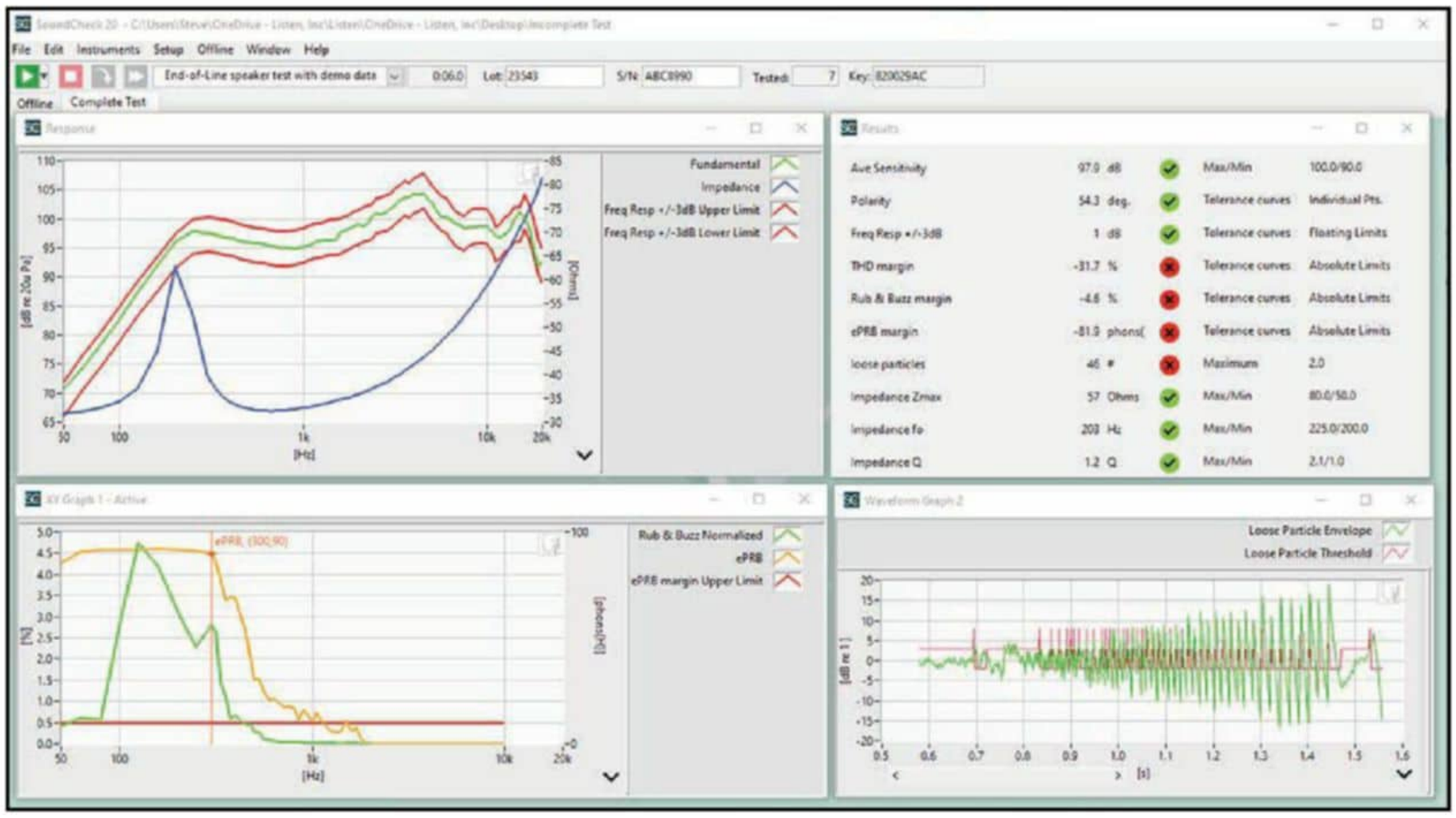


Figure 2: Typical end-of-line tests for a loudspeaker showing frequency response, sensitivity, polarity, THD, Perceptual Rub & Buzz, loose particles, impedance, and resonance

knob is accidentally knocked mid-way through a production run. These potential pitfalls inspired us to develop our own range of audio interfaces which include all the equipment needed for basic audio tests in one chassis with just a single USB connection to the computer.

Products in the AmpConnect family include audio inputs and outputs, microphone power supply, amplifier, impedance measurement, and digital I/O and an amplifier. The smaller, low-cost AudioConnect range offers just the audio interface plus microphone power and a headphone amplifier. All cabling and switching is internal to these devices, preventing cabling errors. Our newer interfaces are also completely free from knobs on the front panel, eliminating any accidental configuration changes; all control is via the test sequence or computer. Naturally, space and cost savings are also achieved by using a single hardware unit rather than separate components.

No matter what combination of hardware and software you choose for your audio test, it is important to ensure that they work well together and that you can calibrate the entire test system as a whole.

Test System Integrity & Specification

The more tightly you specify your tests, the more consistent your end product will be. As I mentioned in the introduction, your production facility aims to produce products that meet your specifications with the highest possible yield and therefore profitability. This means there is little to be gained by testing more thoroughly than dictated by the terms of your contract, and this places the burden on you to be specific. There is a big difference between specifying "frequency response within 3dB of golden unit" and specifying "frequency response within 3dB of golden unit measured with SoundCheck Version 20 and AmpConnect 621 with the provided test sequence in a well controlled test environment."

Without the additional detail in the second option, your manufacturer could be using any test system, which may not measure as accurately as you expect. They may be pairing a software-based system with an inadequate audio interface, for example one that is too noisy or has phase issues that lead to incorrect measurements. Even worse, they might be using a pirated version of a reputable software system! These have a whole host of problems including security issues, older versions of the software without the latest algorithms, look-alike or imitation products that simply do not measure correctly, and of course lack of manufacturer support and upgrades.

At the very least, it is advisable to fully specify the software, down to the current version number, the specific software configuration that is required to run all your tests, and the official authorized vendor from which they must purchase it. Many large OEMs go one step further, purchasing it themselves for their contract manufacturer to guarantee system integrity. If you are relying on your contract manufacturer to purchase the software, it is advisable to ascertain the serial numbers of the systems they plan to use on your lines, and verify their authenticity with the manufacturer.

Hardware is also important. It is critical to use an audio interface with sufficient dynamic and frequency range, and it is also important that it can be fully calibrated with the software. Many times I have witnessed a manufacturer spend good money on a professional audio measurement system, only to attempt to save money by pairing it with a low-cost sound card that is not good enough to get consistent results. It does not need to be the same interface that you use in the lab, as there is often a cheaper alternative that will be adequate for production testing, but do take the advice of your system provider, and ensure that the hardware is also clearly specified in your test plan.

These days, most quality-driven OEMs provide their manufacturer with the specific tests that need to be implemented. With software-based systems, test sequences are easily shared via email,

Test	Reason
Frequency Response	Indicates overall performance of the device. If frequency response is outside the limits, it may affect the tonal balance and will not reproduce sounds as intended.
THD	Indicates overall nonlinear performance of the device. If THD is outside the limits, it may indicate a poorly centered voice coil, or mechanical limitations.
Polarity	A polarity measurement verifies that lead wires are correctly connected and not inverted.
Rub & Buzz	Rub & Buzz defects can be introduced by a number of manufacturing issues such as incorrectly aligned voice coils, incorrectly trimmed lead wires, and more. This simultaneous but separate harmonic analysis can give a detailed insight into the nature of the problem as well as a simple pass/fail result.
Perceptual Rub & Buzz	Measures the audibility of Rub & Buzz based on a perceptual model. This is a great metric to use to decide whether to ship a product as it will reject only those with audible Rub & Buzz, resulting in higher yield.
Loose Particles	Loose particle measurement is a time domain measurement that identifies speakers with small particles trapped under the dust cap, or rattling components such as buttons and fasteners.
Impedance	This electrical measurement checks for resonances, proper magnetization, and Thiele-Small parameters.

Table 1: Typical transducer production tests

network or USB drive. Mindful of the concerns that some OEMs have when sharing their test sequences outside of their direct organizations, SoundCheck now includes additional security features to maintain security and control of the entire test procedure. Measurement sequences can now be locked and protected, preventing operator modification and also ensuring that the sequence cannot even be viewed or opened without a password. This offers complete test confidentiality and security, preventing the test from being modified, copied, or re-created for use with other products. Test sequences can also be locked to specific SoundCheck serial numbers, to ensure that all measurements are being made on legitimate SoundCheck systems.

Factory Integration

Manufacturing environments often require the test system to be integrated with databases and larger factory automation systems. When specifying a test system, consider what data you need to collect and maintain, both now and in the future. Some companies rely entirely on only a pass/fail result, whereas others save all test data to a database or data management system. Many contract manufacturers are now required to save data to the OEMs own database for full traceability. Whatever you need to do, check that your test system offers suitable capabilities and specify how this should be configured.

While production line tests for simple products such as drivers, microspeakers, and earphones tend to be fairly stand-alone, audio tests are often one small part of the overall test plan for more complex devices. In these cases, screen tests, connectivity checks, and other inspections may also take place alongside the audio measurements, and the audio tests must be controlled by a larger test automation system. Every manufacturer implements this differently. SoundCheck uses TCP/IP control, which offers control in any programming language that can provide TCP/IP commands. These commands can not only start and stop tests, but also import and use data stored outside of SoundCheck such as limit curves, start and stop frequencies, and more. While such advanced functionality requires considerable expertise to implement, it does allow professional audio testing with all its advanced techniques and algorithms to be fully integrated into a highly customized test environment.

Operator Instructions & Ease of Use

Document everything! This may seem like common sense, but specific, clearly documented operating instructions, covering everything from initial system configuration, wiring diagrams, and daily system test and calibration directions, to precise instructions about positioning and fixturing, and procedures for a failing test (reject or re-test) will improve your test integrity. If you are providing test sequences to your manufacturers, it's easy to include instructions, either verbal, pictorial, or even video within the test sequence to ensure accurate positioning and other operator interaction during the test, in addition to the stand-alone documentation. Knowledge transfer has been extremely challenging over the last couple of years due to travel restrictions, so it is more important than ever to ensure that all production test operators are adequately trained.

Conclusions

In summary, production testing is complex, and your product quality is only as good as your test system and your system tests. That said, there are several things you can do to ensure a smooth transition:

- Take the time to understand the differences between laboratory and production tests, and what you need to measure.
- Thoroughly evaluate and completely specify the test system for current and potential future needs.
- Set realistic and appropriate production test limits. Know the practical limitations of your test environment (e.g. fixturing and test box), your test system, your manufacturing and product tolerances, and adjust your limits appropriately.
- Clearly document and communicate all test requirements including equipment, tests, environment, and operating instructions. ax

About the Author

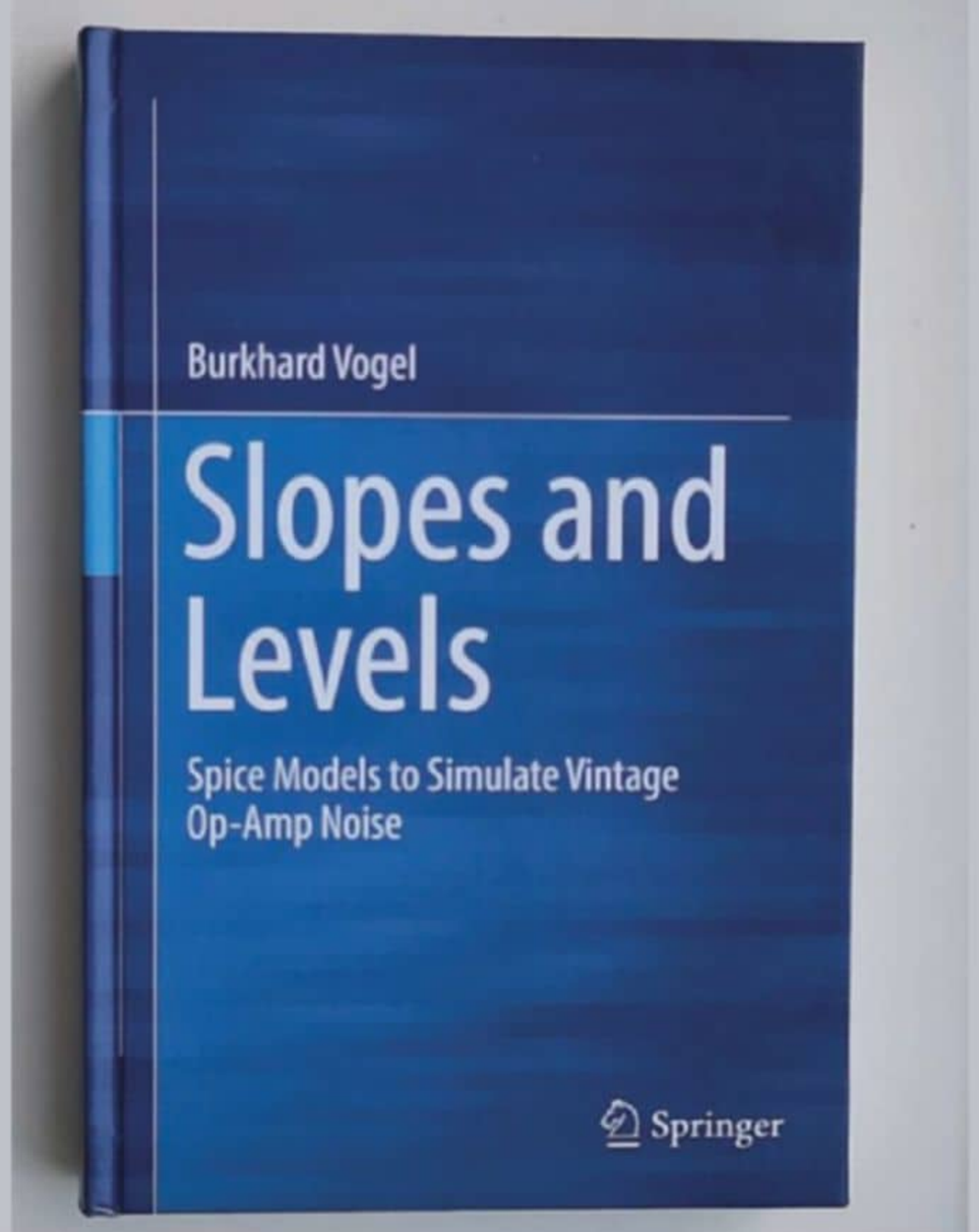
Steve Temme is founder and President of Listen, Inc., manufacturer of the SoundCheck audio test system. Steve founded the company in 1995, and for the past 25 years the company has remained on the cutting edge of research in audio measurement, regularly introducing new measurement techniques, algorithms, and hardware to enable testing of all kinds of devices ranging from basic transducers to complex audio systems. In recent years, Listen's product development focus has been on creating the algorithms, interfaces and test methods to measure today's many and varied wireless and cloud-communicating devices. Prior to founding Listen, Steve worked for many years as an acoustic test and measurement applications engineer at Brüel & Kjær, and also as a loudspeaker design engineer at Apogee Acoustics. He holds a BSME from Tufts University, has authored numerous papers on acoustic testing, and has lectured extensively throughout the world.

Book Review

Burkhard Vogel's *Slopes and Levels*

Spice Models to Simulate Vintage Op-Amp Noise

Review by
Marcel van de Gevel



Interested in accurately simulating the noise of op-amp circuits? Burkhard Vogel wrote a book that addresses this relevant topic for any audio circuit designer who doesn't want to rely just on the manufacturer's own models.

The first question that came to my mind when I read the title of Burkhard Vogel's book *Slopes and Levels: Spice Models to Simulate Vintage Op-Amp Noise* was why such a book would be required, given that op-amp manufacturers nowadays publish op-amp macromodels. Vogel answers that question convincingly: Some manufacturer-supplied models are way off. For example, he found a TL071 model featuring a flat equivalent input noise voltage density of 12 nV/√Hz while a real TL071 is closer to 18 nV/√Hz at 1kHz with a substantial increase below 100Hz. Another example is an NE5534A model with almost 6.5 nV/√Hz of white voltage noise, when a real NE5534A typically produces 3.5 nV/√Hz at 1kHz.

Vogel therefore started making his own LTspice op-amp noise models. He describes in detail what methods he used and what sanity checks he did to see if his models gave the expected results. The methods and equations are easy to understand once you get used to his notation, which doesn't take long, as it is logical and consistent.

The Beginning Chapters

In the first chapters, Vogel tries to use LTspice's UniversalOpamp2 model, but finds it too limited. The model only supports white noise and flicker noise with a slope of -10 dB/decade, while many noise graphs in datasheets show different slopes. On top of that, the noise currents at the positive and negative inputs are always fully correlated with opposite directions, which also doesn't match reality. He therefore switches to a UniversalOpamp2 with switched-off noise and augments it with one noise voltage and two noise current sources. By using Laplace expressions and controlled sources, he can make whatever slopes and correlations are required.

Vogel's approach is strictly a black-box curve-fitting approach. That is unfortunate, because it leads to inaccuracies, some relatively minor ones and one very big one.

For all op-amps, he assumes the current noise to be independent of the voltage noise. This is approximately true for bipolar op-amps and for field-effect transistor (FET) op-amps at low frequencies, but not for FET op-amps in the frequency range where the current noise increases with 20 dB/decade. The low-frequency current noise of FET op-amps is shot noise of junction leakage currents, but the increase at high frequencies is due to the thermal noise in the FET channel in combination with the FET's gate capacitance. As the thermal noise in the channel also causes the equivalent input noise voltage, they are dependent. Fortunately, this only causes an inaccuracy in the total noise when the source has a very high impedance that is largely or completely reactive, such as a condenser microphone capsule.

When trying to find a suitable model for the equivalent input noise current density of an NE5532A, Vogel faces the problem that the datasheets only specify the noise current at 30Hz and at 1kHz, which is not enough to solve the slope of the flicker noise, the level of the white noise and the corner frequency. He rather arbitrarily assumes a -20 dB/decade slope to find a solution that passes through the 30Hz and 1kHz data points, but with a non-

Title: ***Slopes and Levels***
Spice Models to Simulate Vintage Op-Amp Noise
Author: Burkhard Vogel
ASIN: B09ZNYCRZ1
ISBN: 978-3-030-99443-3
Publisher: Springer (May 6, 2022)
Pages: 363

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black-box approach, he might have realized that the equivalent input noise current is predominantly shot noise and flicker noise of the base current of the input differential pair of the NE5532A and that the base current is the input bias current. The typical input bias current of an NE5532A is 200nA, corresponding to white shot noise of:

$$\sqrt{2qI_B} \approx 253.2\,fA/\sqrt{Hz}$$

where I_B is the base current and q the elementary charge. The datasheet specifies 0.7 pA/√Hz at 1kHz and 2.7 pA/√Hz at 30Hz. Subtracting the estimated white noise (well, subtracting the squares and taking the square root) results in the flicker noise at two frequencies. The only -x dB/decade line (where x is some real number) that fits through these points has a -8.074 dB/decade slope and crosses the white noise at 10441Hz. I don't know how accurate or inaccurate these values are, but at least they are based entirely on datasheet values without any need for guessing slopes.

Vogel on Correlated Current Noise

In my opinion, Vogel is incorrect in Chapter 10, "The Correlation Matter," about the correlation between the current noise at the positive and at the negative op-amp inputs, and in Chapter 11, which contains the corresponding Mathcad code. He makes the assumption that whatever correlation between the noise currents is found in the manufacturer's model is the truth, but if the manufacturer's noise models were reliable, there would be no need for the book in the first place.

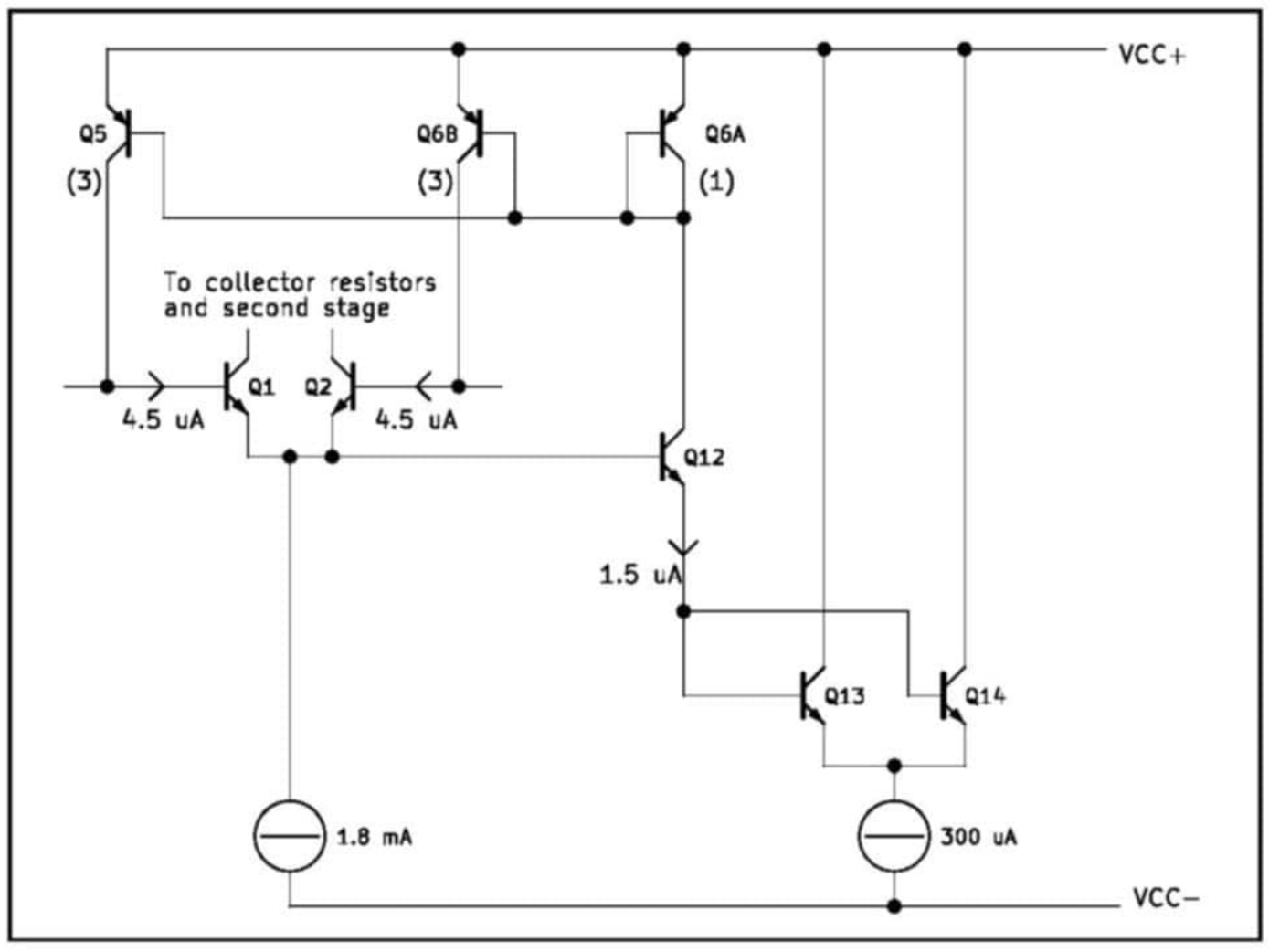

Figure 1: Simplified base current compensation of the LT1128/LT1028

The manufacturer's model of the LT1128 (unity gain stable variant of the LT1028) turns out to have fully correlated input noise currents with opposite directions and a level around 1 pA/√Hz. This contradicts the "Total noise versus unmatched source resistance" graph in the LT1028/LT1128 datasheet, which implies an effective input noise current just above 3 pA/√Hz when the impedances driving the positive and negative inputs are very different. It also contradicts noise current measurements of Paul Horowitz and Winfield Hill, authors of *The Art of Electronics*, who Vogel quotes in his Figure 14.16. These show about 4 pA/√Hz. What it all boils down to, is that the Linear Technology model only models the differential component of the input noise currents and not the much larger common-mode component.

Following Linear Technology's inadequate noise model for the LT1128, Vogel makes a model with the same deficiencies and even conjectures that op-amps with base current compensation have fully correlated input noise currents with opposite directions. This would be true if base current compensation circuits sensed the sum of the momentary positive and negative input bias currents and corrected for those, but as far as I know, such base current compensation circuits have not been invented yet. Real base current compensation circuits use one or more transistors that are a (scaled) copy of the input transistors and mirror out its/their base current. As the copies closely match the input transistors, this results in the correct DC compensation current. It doesn't compensate for the common-mode component of the input noise current, because the scaled copies generate shot noise independently of the input transistors—in some implementations, the common-mode component of the input noise current even gets much larger, meaning that the noise currents of the positive and negative inputs are strongly correlated with equal rather than opposite directions.

Base Current Compensation Example

My **Figure 1** shows a simplified schematic of the base current compensation circuit of an LT1028 or LT1128. Several cascodes, helper transistors and a trick to ensure that Q_{13} and Q_{14} have the same collector-emitter voltages as Q_1 and Q_2 have been left out for simplicity, the full circuit can be found in the LT1028/LT1128 datasheet. Transistors Q_1 and Q_2 are the input differential pair. Q_{13} and Q_{14} are the model transistors, which are biased at a six times lower current; these transistors are presumably also six times smaller than the input transistors. The sum of the base currents of Q_{13} and Q_{14} is mirrored and amplified by current mirror Q_{6A}-Q_{6B}-Q_5. The

mirrored and amplified currents are connected to the inputs to compensate for the base currents of the input pair.

First looking at the parts that are not common to both inputs: with their base currents of 4.5µA, the input differential pair transistors each have a base shot noise $\sqrt{2qI_B}$ of 1.2 pA/√Hz. Transistors Q_{6B} and Q_5 each contribute 1.2 pA/√Hz of collector shot noise. If these were the only noise sources, we could add the squares and take the square root, and end up with uncorrelated input noise currents of 1.7 pA/√Hz at the positive and 1.7 pA/√Hz at the negative input. Mathematically, you can split this into a common-mode noise current of 1.2 pA/√Hz per input and a differential noise current of 1.2 pA/√Hz, which is reasonably close to the specified 1 pA/√Hz differential noise current. So far, so good.

Now looking at the common parts: Q_{13} and Q_{14} together have a base current of 1.5µA and together produce 0.693 pA/√Hz of base shot noise. Q_{12} does almost nothing and Q_{6A} produces another 0.693 pA/√Hz of collector shot noise. Taking the root of the sum of the squares, the total is 0.98 pA/√Hz. The amplifying current mirror amplifies this by three and injects it into the positive input and into the negative input, completely correlated with the same direction. Adding the 1.2 pA/√Hz of common-mode noise current that we already had, the total common-mode noise current becomes 3.177 pA/√Hz. When one input is driven from a very low impedance and the other from a not so low impedance, which is the normal use case for a low-noise op-amp since you normally keep the feedback network impedance very low, both the common-mode and the differential-mode noise currents come into play and the total noise current is 3.396 pA/√Hz, which is in between the values from the "Total noise versus unmatched source resistance" graph shown in the LT1028/LT1128 datasheet and the Horowitz and Hill data.

A minor mistake in Chapter 20 about the OPA134 is Vogel's interpretation of the simulation graphs 20.15 and 20.16. These don't show a weird bug in the noise model, but simply that the manufacturer has included about 4pF of capacitance from each input pin to the supplies in the macromodel.

Overall Impression

While Burkhard Vogel's *Slopes and Levels* may not be the last word yet, it is definitely a valuable resource for anyone interested in accurately simulating the noise of op-amp circuits.

About the Author

When **Marcel van de Gevel** was 9 years old, he got a Philips electronics kit. He has been an electronics hobbyist ever since, with a particular interest in audio circuitry. He also turned electronics into his profession: in 1993, he got an ingenieur's (MSc) degree (cum laude) in electronics engineering from the Delft University of Technology. Professionally, he mainly works on RF integrated circuits for consumer equipment.

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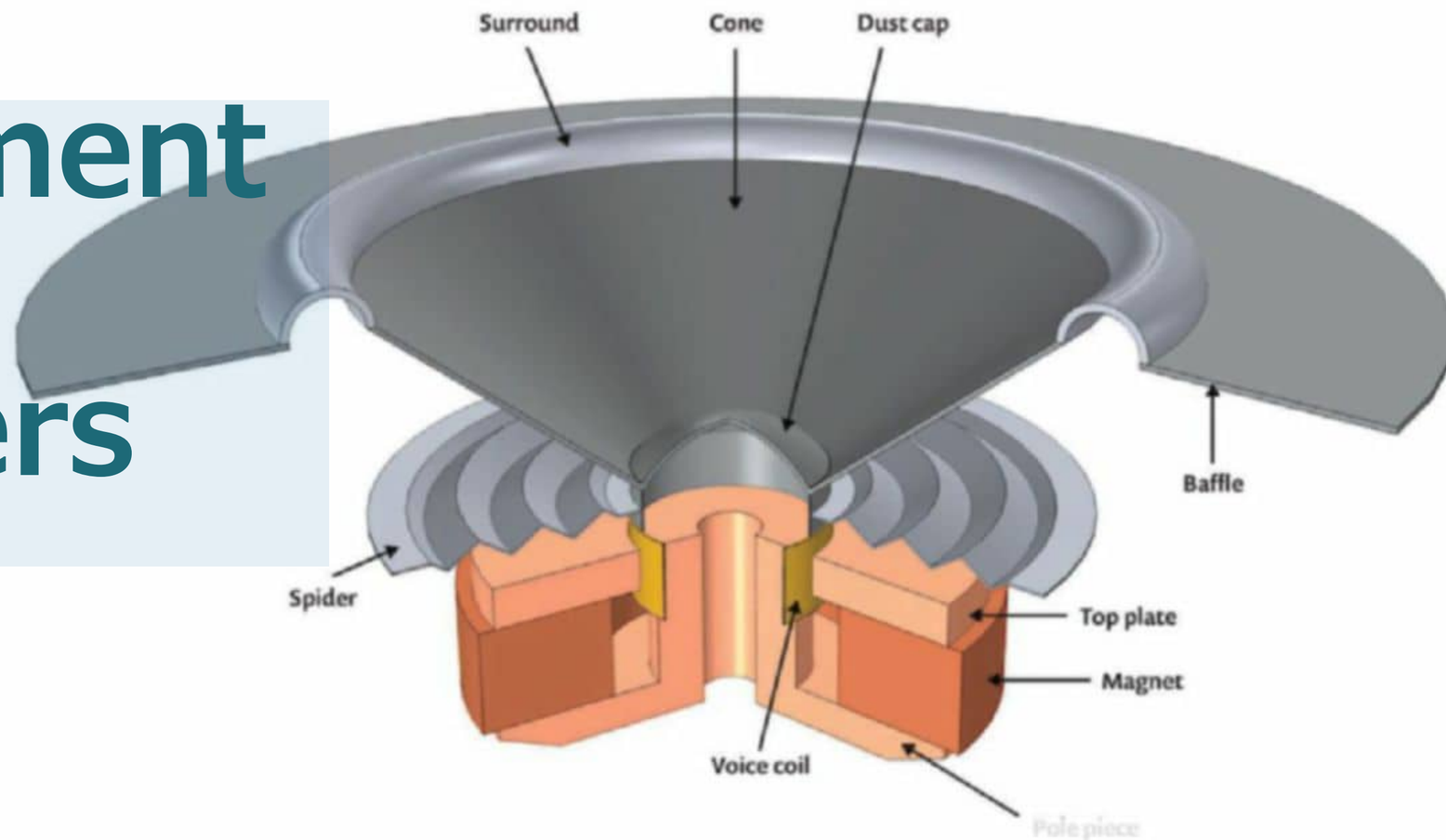
Lumped Element Modeling of Transducers



By
René Christensen

Many engineering tasks require a common ground from which discussions among those involved in a project can take place in a structured manner. This will often involve a so-called model of the concept or product, and this article discusses one such model: the lumped element model.

Lumped element modeling involves representing continuum-physics such as a loudspeaker with various three-dimensional parts as discrete or lumped zero-dimensional components and make so-called analogy circuits that to some degree capture the physics of the transducer involved. The crux of making these analogy circuits is to realize that there is an equivalence such that the equations relating the primary and secondary variables in one physics are analogous to the equivalent equations in other physics.

The impedance in each physics domain gives you the "primary" or "independent" variable divided by "secondary" or "dependent" variable, which for the electrical domain is the voltage divided by the current. The electrical domain serves as a "parent" domain, since we want to take advantage of the well-established framework here with Kirchhoff's current and voltage laws, establishing transfer functions from circuits, Bode plots for frequency responses, using SPICE software, and so on. The downside of course being that an engineer trained purely in acoustics or mechanics might not be immediately comfortable with these circuits. We see in **Table 1** how impedance is defined in the different domains and notice the similarities.

So, one can find an analogy between an electrical capacitance and a compliance, be it mechanical or acoustical, and similarly, between an electrical inductance and a mass. We will see some complex circuits later in this article utilizing this mapping, but first imagine the situation shown in **Figure 1** with two springs being compressed by a common force. It should be easy enough to realize that the total stiffness will be the sum of the stiffness of each of the springs on their own. As stiffness is the inverse of compliance, we can thus calculate the total compliance as:

$$C_{m,total} = \frac{1}{\frac{1}{C_{m,1}} + \frac{1}{C_{m,2}}}$$

And here we see an issue. From electrical engineering we recognize this as the way to add capacitance when in series in an electrical circuit, whereas for the mechanical setup we would immediately say that the two springs are in parallel. Figure 1 illustrates a well-known issue being that you will not automatically have a "conform" depiction when mapping from one domain (such as mechanical) to another (electrical).

To have congruence with respect to series vs. parallel connections between two non-conforming domains, one can instead adapt the so-called Admittance or Mobility strategy, where impedances in one domain have admittance (inverse of impedance) analogs in the other, so that the primary variable in one domain maps to the secondary variable in the other domain, and vice versa.

The admittance analogy will ensure that series and parallel connections are retained in the electrical domain, but as with anything, there are pros and cons to both the impedance analogy approach and the admittance ditto. I typically work consistently with the impedance analogy, but sometimes the analogy circuits are much easier to make in the admittance analogy first, and then subsequently

I convert to the impedance analogy. The choice will also be affected by the transductance principle for the transducer, meaning how energy is converted between domains, which is different between a standard "electrodynamic" driver and for an electrostatic transducer as examples, and we may touch more upon this in a future article. Finally, it should be noted that special cases exist for which only one of the two approaches is possible.

Degrees of Freedom

One important aspect of a lumped model is that it has a limited number of degrees of freedom, so there is only so much of the continuum physics aspects of a real transducer that are captured. Oftentimes it is not the lumped aspect itself that limits the information that can be extracted from a model, but instead the number of components that were put into the model in the first place.

For example, looking at the typical lumped circuit for a loudspeaker, as shown in **Figure 2**, utilizing the Impedance analogy strategy throughout, it is seen that all the moving mass of the loudspeaker resides in only a single inductor, and the total stiffness from primarily the spider and the surround sits in a single capacitor. This limits the effects that would be seen in a more elaborate lumped model, where the basket was included as perhaps two added compliances between the magnet and the spider, and the spider and the surround/enclosure, respectively, with each their associated mass, to capture via the lumped model how the reaction force in the magnet systems displaces the individual parts of the driver differently.

Note however, that in Figure 2 a component has been added for the mass of the cabinet and magnet system. The reaction force will displace the magnet in the opposite direction of the voice coil, and while the effect is small for a traditional loudspeaker, we will see later that for some applications it should not be ignored.

Balanced Armature Receivers

One could go straight to focusing on having a very rigid basket, such as has been sought by, for example, Børrensen [1] using topology optimization (a topic for a later article), see **Figure 3**, along with a heavy and stiff cabinet to have the physical setup match the simpler traditional lumped model for a loudspeaker. But I think it is a good idea to have more components in the initial model, and then from there estimate where to stiffen the basket, or in general investigate the relevant variables in a model complex enough that it does not lump several of them together.

	Electrical Domain	Mechanical Domain	Acoustical Domain
General Impedance	$Z_e = \frac{u}{i}$	$Z_m = \frac{f}{v}$	$Z_a = \frac{p}{q}$
Resistance	$Z_{e,\mathrm{R_e}} = \mathrm{R_e}$	$Z_{m,\mathrm{R_m}} = \mathrm{R}_m$	$Z_{a,\mathrm{R_a}} = \mathrm{R}_a$
Capacitance/ Compliance	$Z_{e,\mathrm{C_e}} = \frac{1}{isC_e}$	$Z_{m,\mathrm{C_m}} = \frac{1}{isC_m}$	$Z_{a,\mathrm{C_a}} = \frac{1}{isC_a}$
Inductance/Mass	$Z_{e,\mathrm{L_e}} = sL_e$	$Z_{m,\mathrm{M_m}} = sM_m$	$Z_{m,\mathrm{M_a}} = sM_a$

Table 1: The impedance for electrical components and how analog expressions are found in mechanical and acoustical domains

While the basket and the cabinet of a loudspeaker certainly does flex and move due to reaction forces, there are transducers for which this effect is much more important, and one such transducer is the Balanced Armature Receiver (BAR). In **Figure 4** a cross-section of the BAR is shown for a so-called Single version since it has a single membrane. The armature is magnetized by an AC current in the coil, and as the armature is suspended between magnets, a displacement can be obtained via this "moving iron" setup.

A principle lumped model can now be presented here that captures the relevant physics; the electrical behavior, the mechanical behavior, and of course the most important of them, the acoustical outputs, in the respective domains represented by the lumped components. The model considers the effect of reaction forces possibly displacing the whole receiver housing and how this modifies the acoustic pressures. The sequential coupling from

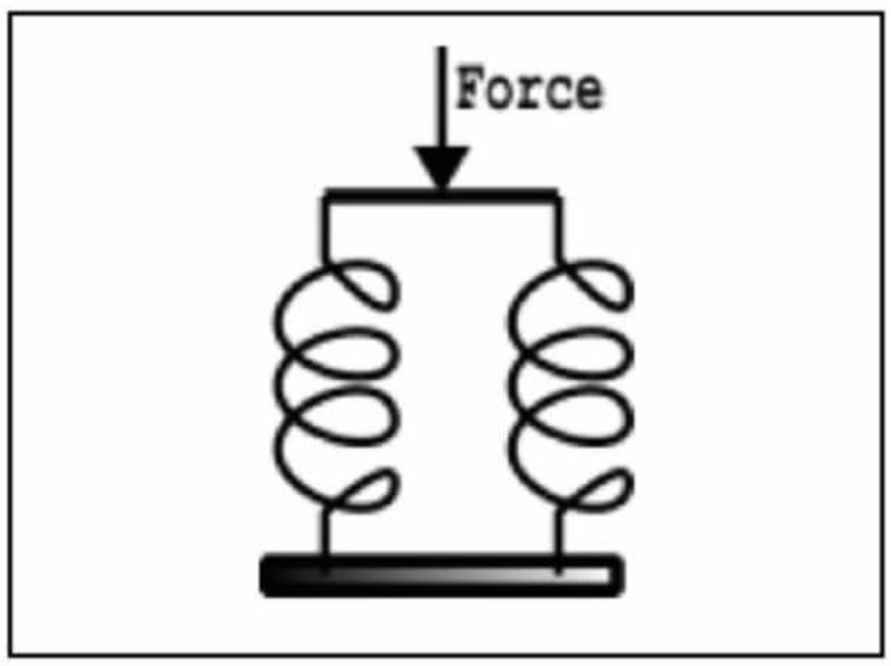


Figure 1: Two springs mechanically in parallel

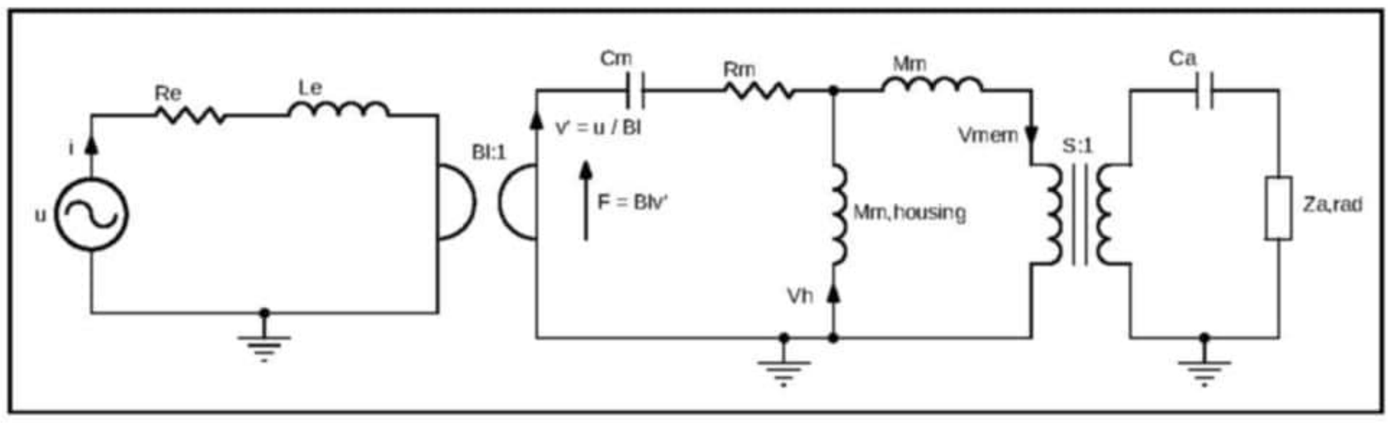


Figure 2: Analogy circuit for loudspeaker driver in a closed and mobile enclosure

electrical to mechanical is two-way, and the same goes for the mechanical-acoustical coupling. This means that one could investigate the induced voltage and acoustic pressure resulting from vibrating the housing, and that could be of interest to hearing aid manufacturers and earphone companies. The lumped model is shown in **Figure 5** with the individual components for each relevant part of the Single BAR.

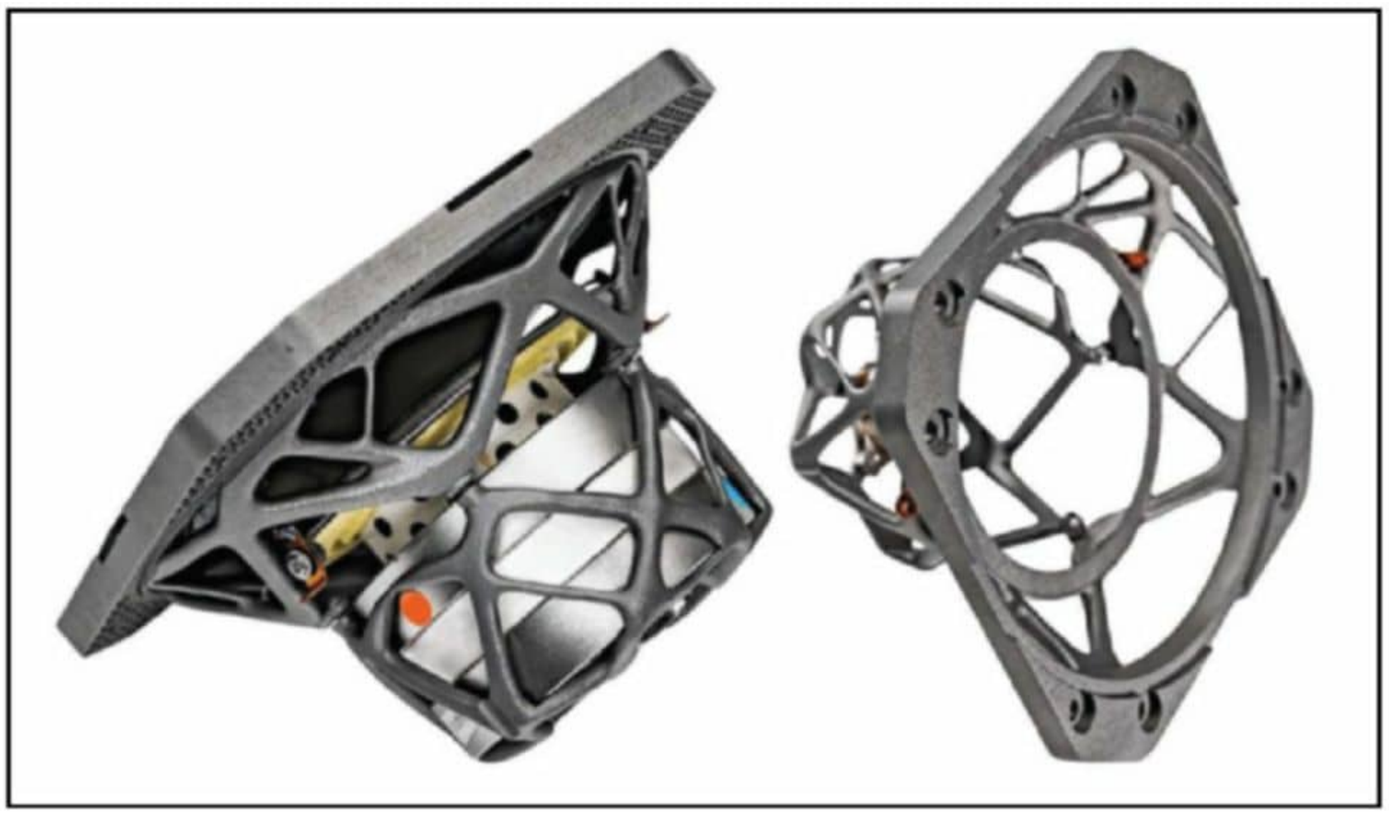


Figure 3: Børrensen's mechanical topology optimized basket for the M1 bass/midrange unit [1]

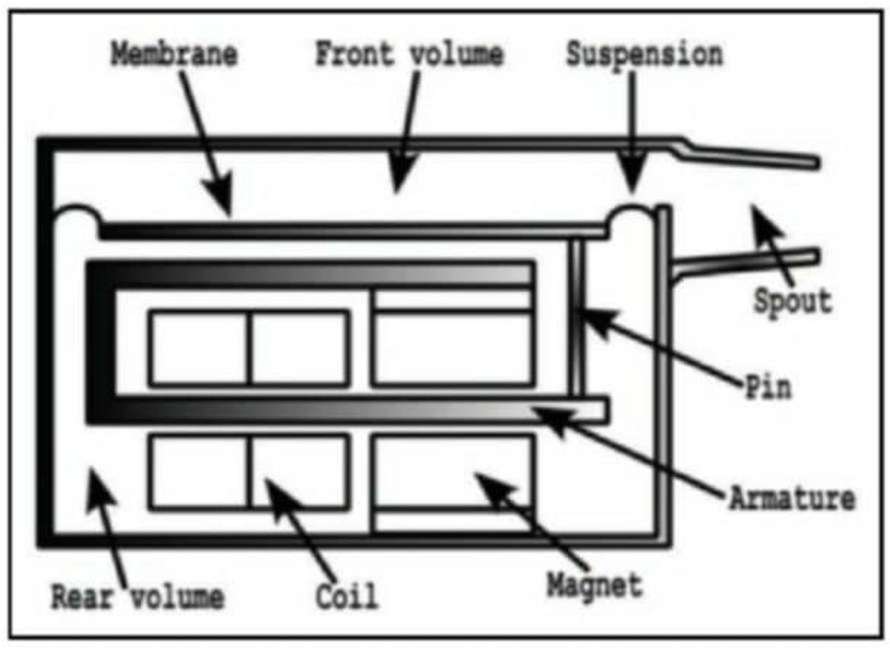


Figure 4: The internal build of Single BAR with a U-shaped armature

The lumped circuit for the Single was made in Spring 2022 and components were fitted up against an existing finite element of the Single BAR made by COMSOL and can be found in its Application Library [2]. Note that all fitting here was "physical" in that component values were found via geometry and material parameters, instead of simply doing a numerical fitting.

The agreement between the lumped model and the finite element model is good across all involved physics up to a frequency where lumped modeling requires more degrees of freedom to capture the relevant effects, with more details on my own blog post on the model [3], and with the sound pressure levels for the two models compared in **Figure 6**.

With such a model, the output from more complex setups can be evaluated quickly, such as for the Shure SE535 earphone shown in **Figure 7**. The earphone utilizes two BARs—a Single and a Dual—and it incorporates a crossover filter between the two. Their combined output into, for example, a front volume, a tube, and a coupler, can now be evaluated in a matter of a few seconds, and the model will encapsulate aspects such as complex electrical impedance and complex velocities for each of the masses. Design changes are easily made via the available parameters for materials and geometries, and so a parameter sweep or even parameter optimization can be used for exploring the solution space.

With the combination of a Single and a Dual, one now of course has the option to optimize each of them for their intended output with respect to their individual frequency responses, instead of having the usual compromises found for a lonely receiver, Single or Dual, found for example in a hearing aid. With two different receivers in the earphone, the Single BAR can now be modified to extend further up in frequency, whereas the Dual BAR can handle the lower frequencies on its own. The block diagram

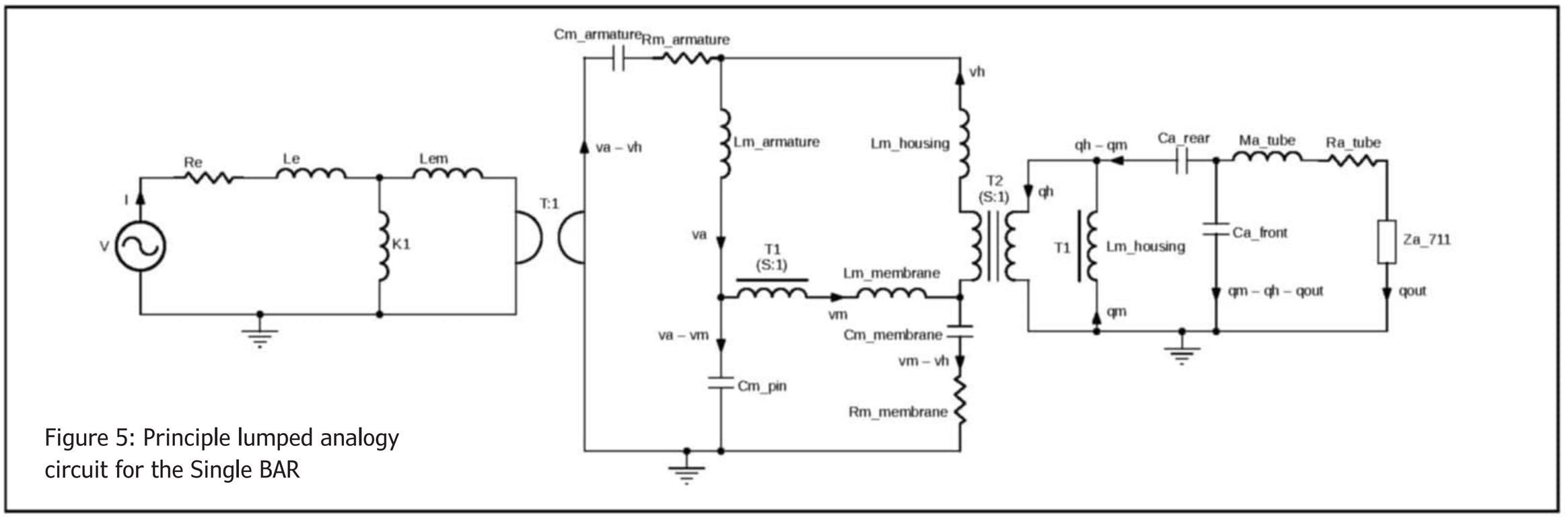


Figure 5: Principle lumped analogy circuit for the Single BAR

shown in **Figure 8** outlines how the different lumped sections can be connected for evaluating the coupler pressure resulting from an applied input voltage.

Circuit Calculations

The transfer functions for the crossovers could be something like a second order Linkwitz-Riley type (of course I have no idea what type of filter is being used), and so mathematically it would be expressed as:

$$H(s) = H_{lo}(s) + H_{hi}(s) = \frac{-\omega_0^2}{s^2 + 2\omega_0 s + \omega_0^2} + \frac{s}{s^2 + 2\omega_0 s + \omega_0^2}$$

and this could be incorporated purely mathematically into the overall circuit calculations, which would be equivalent to having a so-called active crossover, or it could be realized via a passive implementation, so that each filter section is realized with explicit components that then connect to the components for the Single and the Dual lumped circuits, respectively. The latter setup would correctly capture the effect of the receivers loading the filter section, and so the designers could easily try out different filter configurations.

For the Dual BAR the lumped circuit is seemingly missing, and so one could go by way of creating a full finite element model in COMSOL Multiphysics as shown in **Figure 9**. However, I have also made

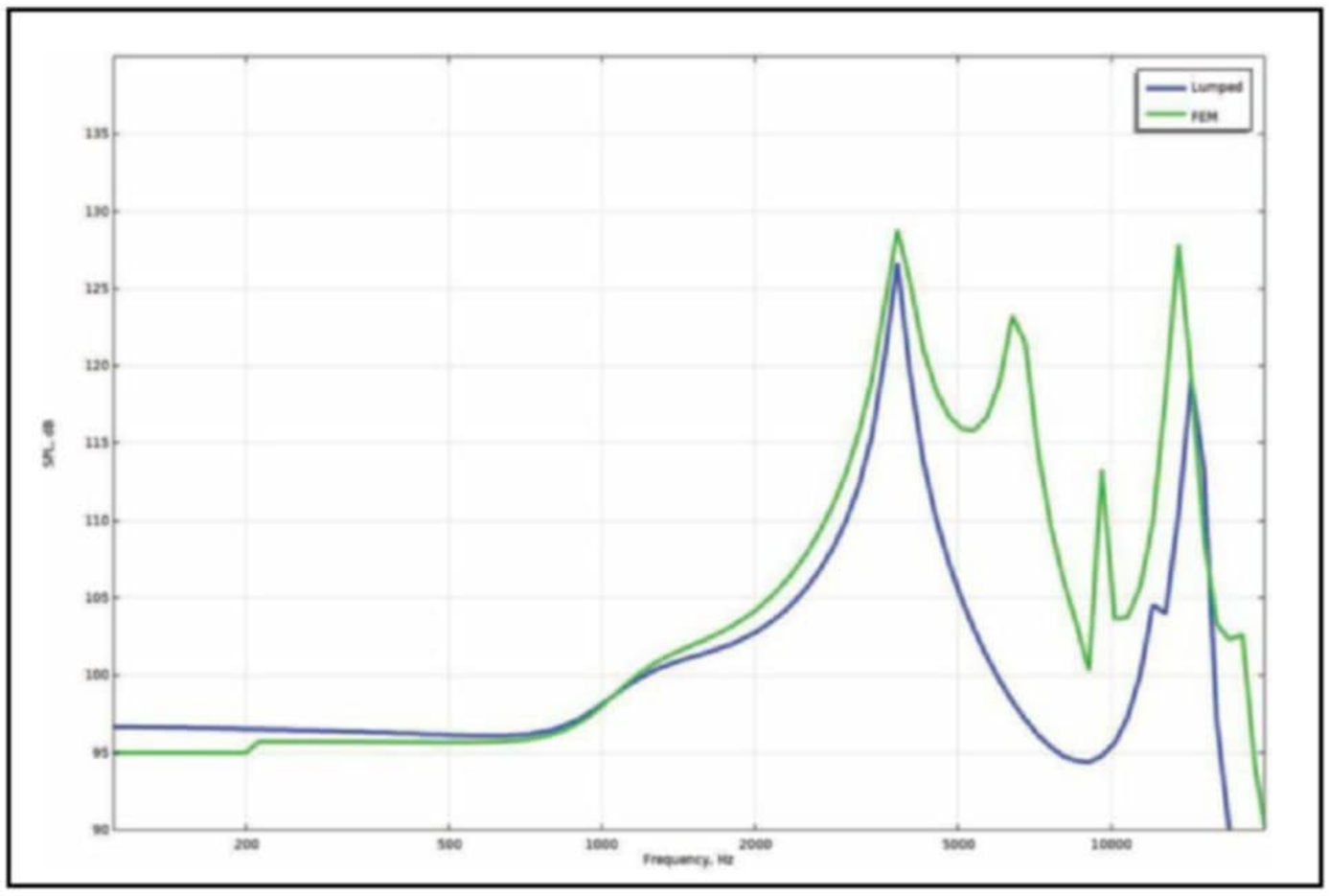


Figure 6: Sound pressure level in the coupler for the lumped model and the finite element model

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the Dual BAR lumped circuit, which is presented here for the first time in **Figure 10**. Again, there is very good agreement between the two models, and it is my hope that my new lumped models will benefit transducer and hearing aid engineers in their development work.

The presented circuits have complete couplings between each intersection, and perhaps that can lead to new utilization ideas, such as counting steps based on the induced voltage from the housing moving. These circuits are very powerful for transducer engineers, since once they are established, a lumped model runs in about one second, while the more complex finite element models can have computation times that can be three, four, or even five orders of magnitude longer. That means that one should start out with the lumped model for the overall estimates and parameter studies, and then continue with the finite element model for the refinements at higher frequencies as needed.

Figure 7: Exploded view of the Shure SE535 earphone. Thanks to Gojko Obradović for suggesting this Shure earphone as an example.

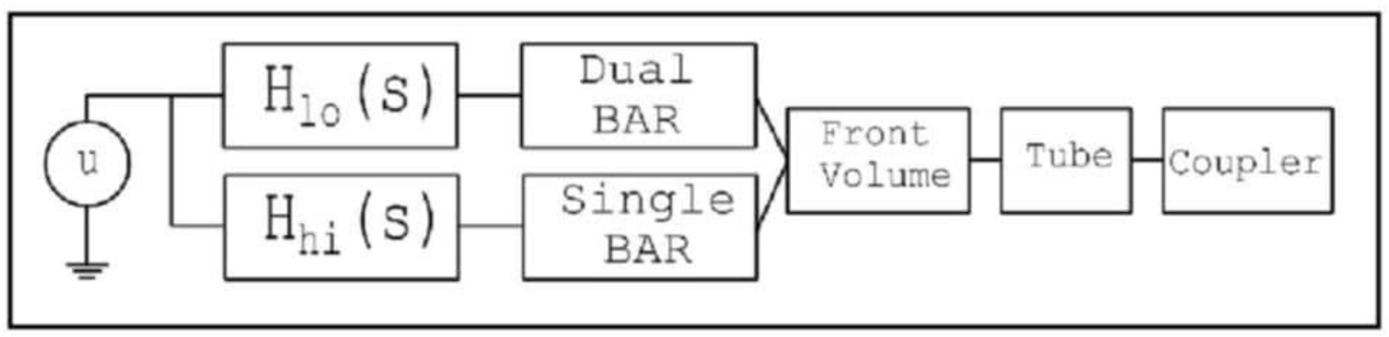
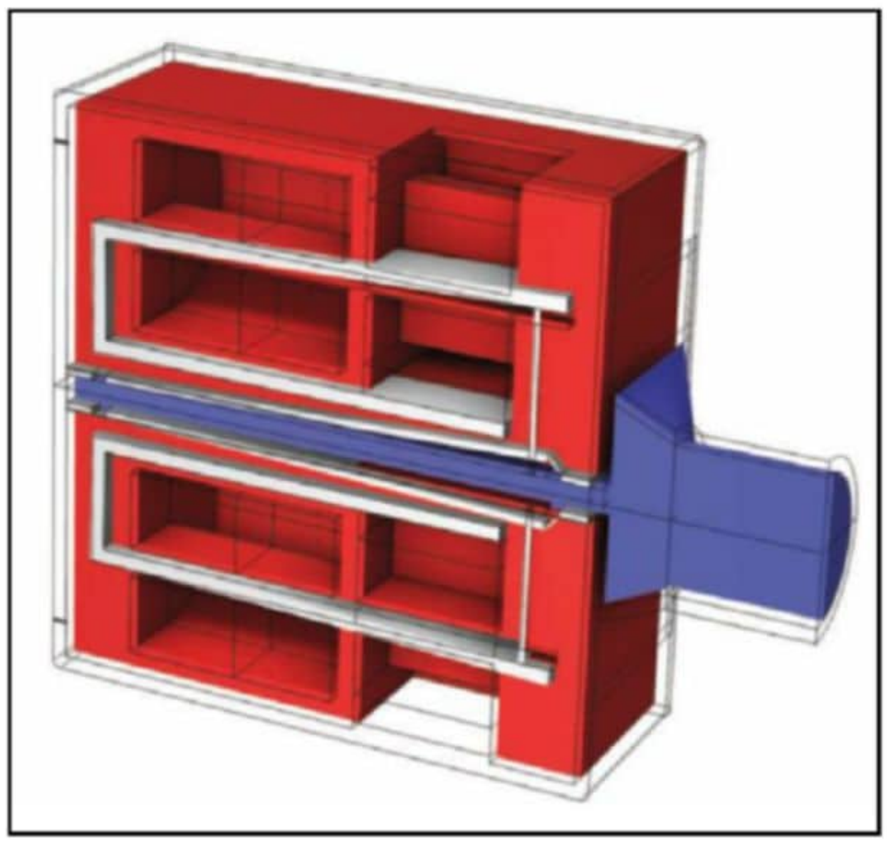


Figure 8: Block diagram for connecting the lumped element circuits

Figure 9: Finite element model for the Dual BAR with an exaggerated view of the displacement



Concluding Remarks

I am a big proponent of modeling-based engineering. Not any one modeling type in particular; it could be analytical expressions, lumped models, transmission line models, finite element models, or hybrid models that combine the aforementioned types. For me, it is very difficult to discuss designs and inner workings of a product without any type of model characteristics. The lumped modeling method described in this article can often give engineers a very good representation of the product that they are working on, and it helps immensely in meetings to have something concrete to present and discuss. The lumped model type, like any other, has its limitations, but it is often the best place to start when starting out on a new design, and as has been demonstrated, you can cover some complex designs and still have an extremely quick design change cycle.

In one of my next articles, we will look at a client case where a transducer setup was modelled using the lumped approach, and how the client can do some of their design explorations via delivered analogy circuits.

Author Acknowledgements: Thanks to Gojko Obradović (Principal Hardware Architect Acoustics, Oticon A/S) for suggesting the Shure earphone as an example.

Resources

[1] “Børrenson M1: New Flagship Loudspeaker from the Danish Company,” Fidelity Magazine.com, www.fidelity-magazine.com/borresen-m1

[2] “Balanced Armature Transducer, Comsol, www.comsol.com/model/balanced-armature-transducer-61741.

[3] R. Christensen, “#40: Balanced Armature Receiver - Lumped Parameter Modeling (Preprint, Part 1), June 11, 2022, www.acculution.com/single-post/040-balanced-armarture-receiver-lumped-parameter-modelling-preprint-part-1

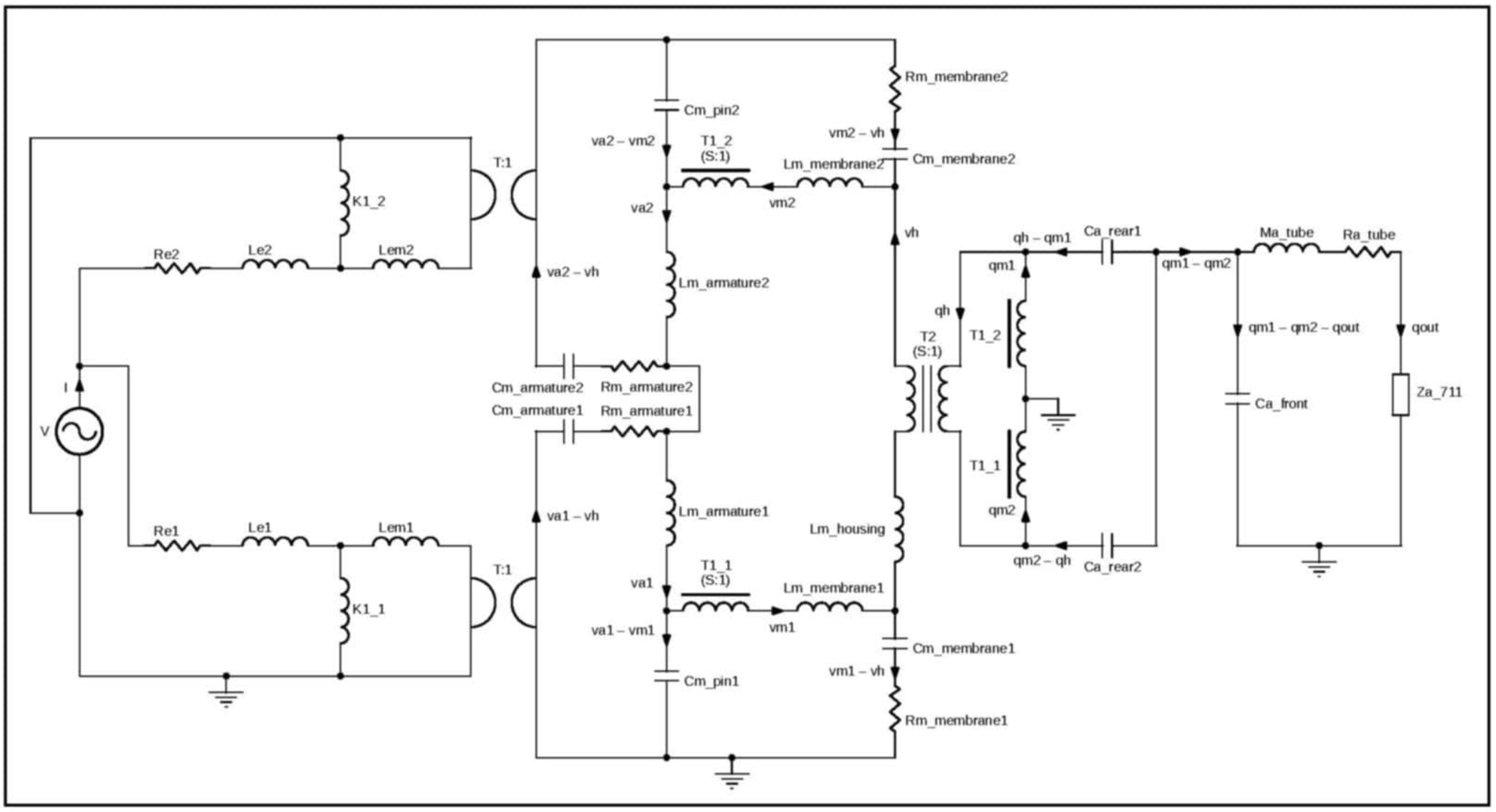


Figure 10: The principle lumped model for the Dual Balanced Armature Receiver

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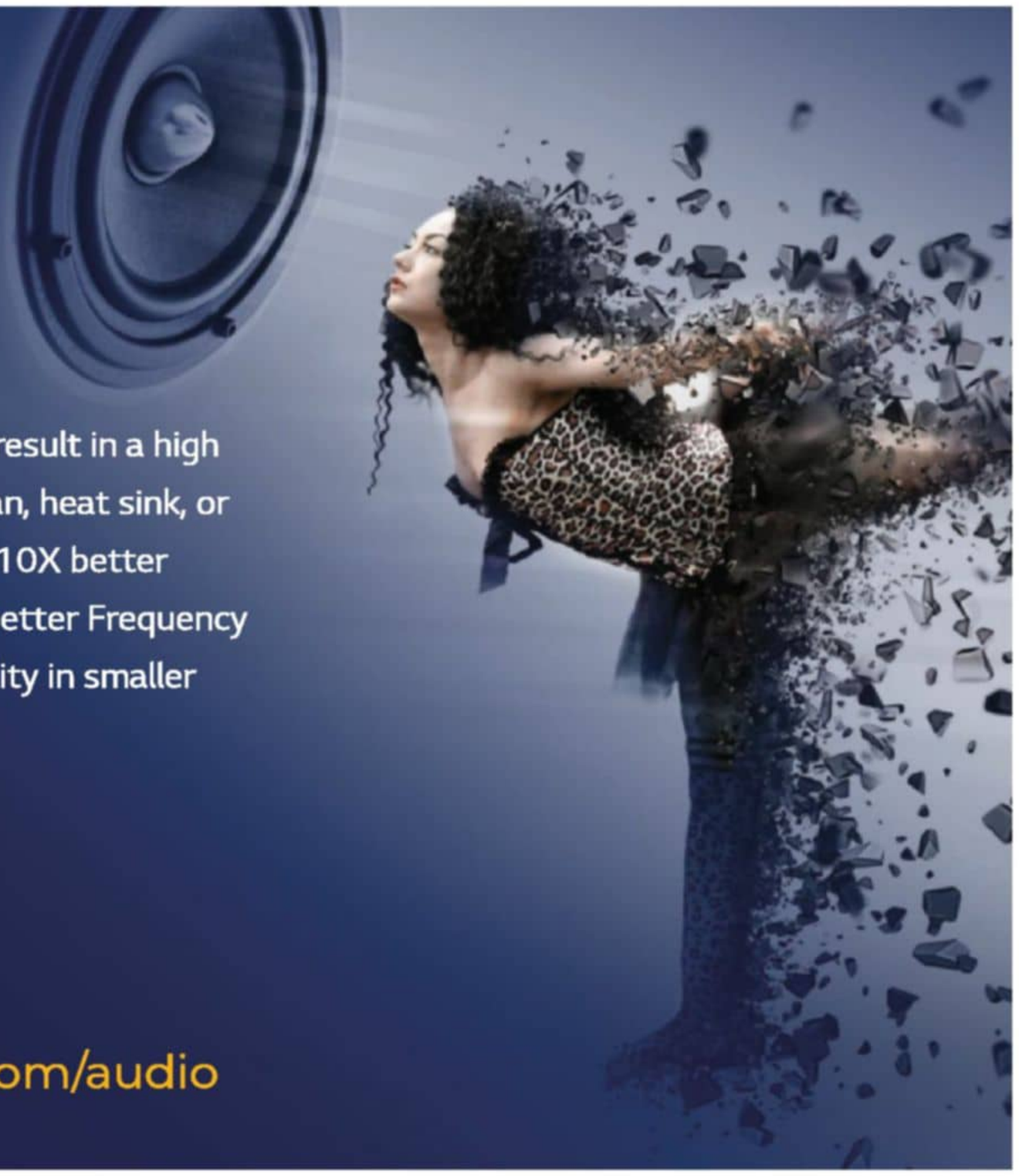
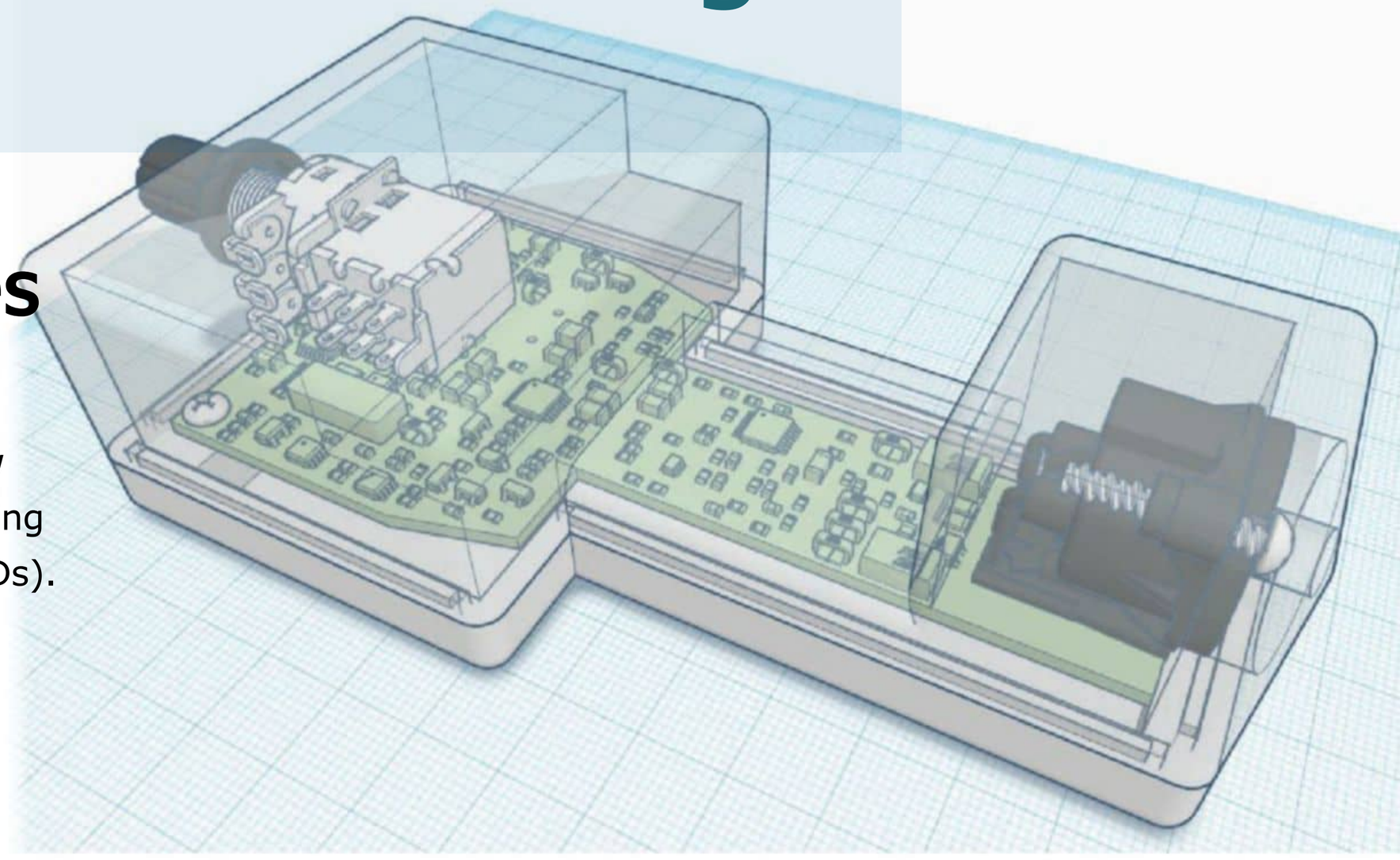


Modern PC Board Design Practices

Part 2 — Tips and Techniques

In Part 1 of this article Ethan Winer gave a general overview of modern PC board design using surface-mounted devices (SMDs). Part 2 continues with specific techniques and advice.

By **Ethan Winer**



Engineer Julia Truchsess designed and 3D-printed this prototype microphone case.

As explained in Part 1, unless your electronic device is fairly simple, a four-layer board makes sense because it offers dedicated ground and power planes in the middle, as well as a bottom layer for additional traces or power. This also avoids having to run both signal and power traces on the same layer that might complicate routing. Further, the inner layers avoid capacitive coupling between top and bottom traces that could, for example, cause a high-gain op-amp to oscillate if its input and output traces are opposite each other for much of a distance.

The top layer holds the component pads and most of the connecting traces, with the inner ground layer beneath. Then another inner layer carries power around the board, with the bottom layer available for any additional traces that can't fit on top. To my thinking, laying out the routing of a PC board is somewhat like playing chess. You not only have to solve the current problem of how to connect component A to component B, you also have to think ahead several moves: If I put R35 here, will that block pin 6 of the op-amp from connecting to C12 over there? Would it help if I rotate the dual NOR gate 180 degrees?

As **Figure 8** shows, besides laying out the board traces, it's just as important that the component labels (R12, C14 etc) on the silk screen layer clearly identify their related parts. Especially with tightly packed boards like my current project.

Photo 6 shows the completed board. I try to orient the text labels to align with their parts, but connecting lines were needed in a few places such as C33 near the middle of the board. The solid green-blue circles are holes through the board. The three large holes are used to align the board with the solder mask template when applying the solder paste, and also for later mounting in the case. The two small holes at right accept little nubs on the XLR connector to keep it aligned.

It's fine for labels to cover traces, and even vias if needed, because those are coated with a clear protective film and don't receive solder. Though avoid the holes in vias, or position letters such as "C" and "O" to circle the hole. KiCad uses short and long lines to identify ICs, with the longer line next to Pin 1. But I added small diamonds as text to mark Pin 1 even more clearly. I also added arrows to all the diodes to better identify their cathodes, and plus signs for the polarized capacitors to augment the too-small circles KiCad uses. Of course, a mass-produced board that's assembled by machine doesn't need these labels, so the parts layout could be even tighter.

That leaves the bottom layer shown in **Figure 9** free for traces that need to cross traces on the top layer. Other than these few traces, the entire bottom surface is filled with copper. This circuit runs on a single 5V supply, so the bottom layer's fill carries V/2 (2.5V) as a "center" pseudo-ground reference for

the audio signals. Both KiCad and ExpressPCB create the fill clearance areas around traces automatically, though KiCad does a better job. With ExpressPCB you often need to draw a “keep-out” manually if the fill goes too close to a trace or via, or if a filled area ends up too narrow.

Also notice in Figure 9 that there are two types of vias that connect to traces on the top layer, shown as white circles. Vias without a black clearance around them connect this V/2 layer to circuit traces on the top. Vias with a clearance connect traces on top to one of the inner layers, or to traces on this bottom layer. Pads for leaded (through-hole) components or wires that connect to a copper fill can be either solid or “thermal relief” as shown in **Figure 10**. The pad is still connected to the copper fill, but having less copper makes it easier to manually solder the component leads quickly without the fill acting as a heatsink.

Fills are also used on the top layer when the layout isn’t as dense as this one. In that case the filled areas are connected to ground to minimize capacitive coupling between adjacent traces. It’s also possible to have SMD components and their pads on the bottom of the board. Though if you’re using solder paste and an oven, you might want to solder components on the bottom manually, after the top parts have been baked in place. I never tried using an oven a second time with the board upside down for parts on the bottom. But my friend Julia Truchsess insists that parts on top won’t fall off in the oven unless they’re large and heavy. The same surface tension mentioned earlier in Part 1 that slides the parts into alignment also holds them in place on the board when they’re upside down!

While we’re on the subject of soldering, it’s worth adding that solder flux is partially conductive. The standard DIY way to clean a PC board after the parts are soldered is to dip the board in isopropyl alcohol and clean it gently with a soft brush. Low impedance circuits can often get away without cleaning, and some types of solder paste (e.g., Chip Quik SMD291AX series) don’t require cleaning at all. Further, using solder paste with a template mask and oven doesn’t spread flux around the board as much as hand soldering does. But if you have any resistors in the megohm range it’s usually wise to clean the board.

Running Traces

Use appropriate trace widths and trace spacings based on the amount of current (width) and voltage (space) being carried. Of course, the board’s copper thickness is also a factor for how much current a trace can handle. With most modern solid-state

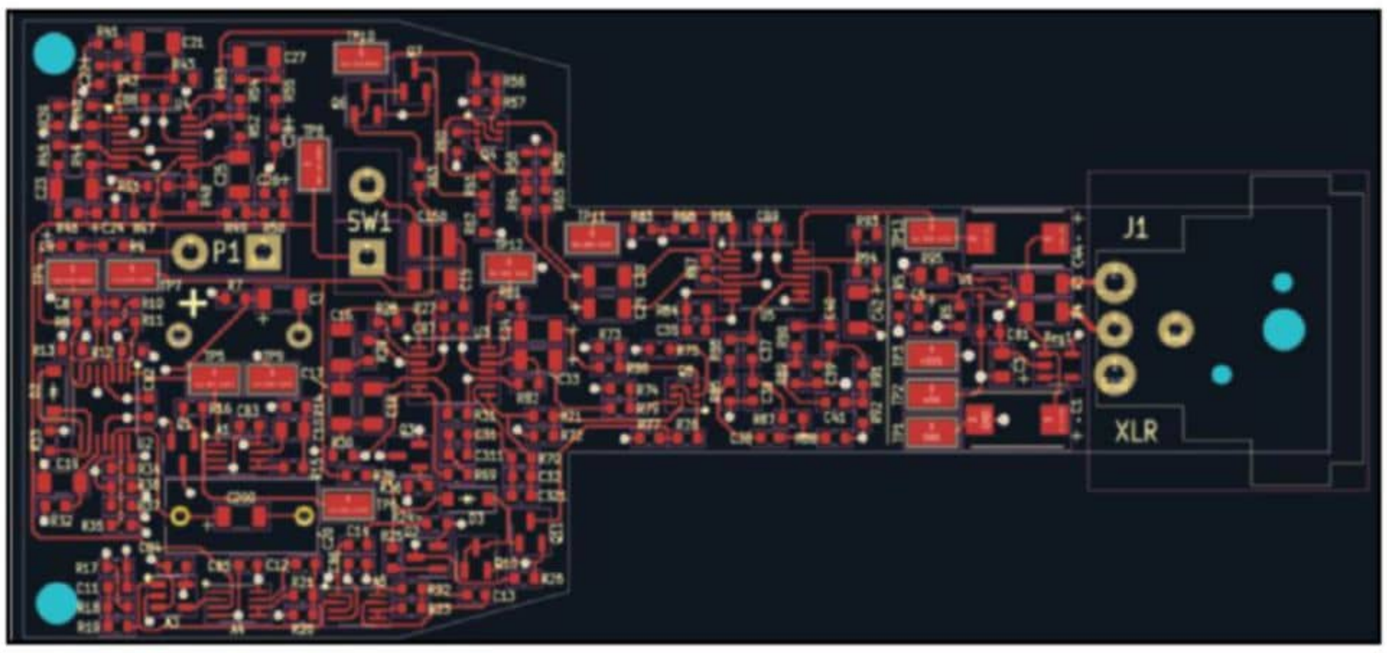


Figure 8: The top copper layer as seen in KiCad holds the solder pads needed for every SMD component, along with the traces that connect them.

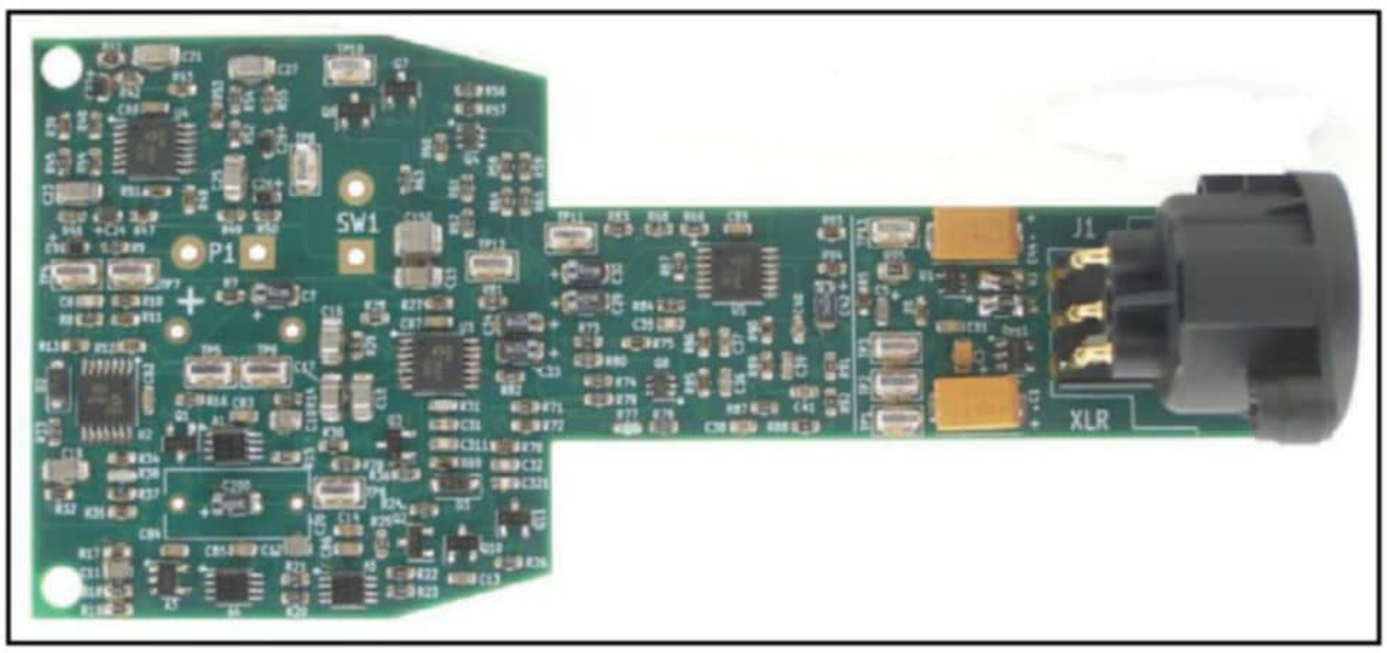


Photo 6: This is the finished board in all its glory. With nearly 200 parts packed so tightly, there’s no room for a solid copper fill between the traces as on the bottom layer in Figure 9.

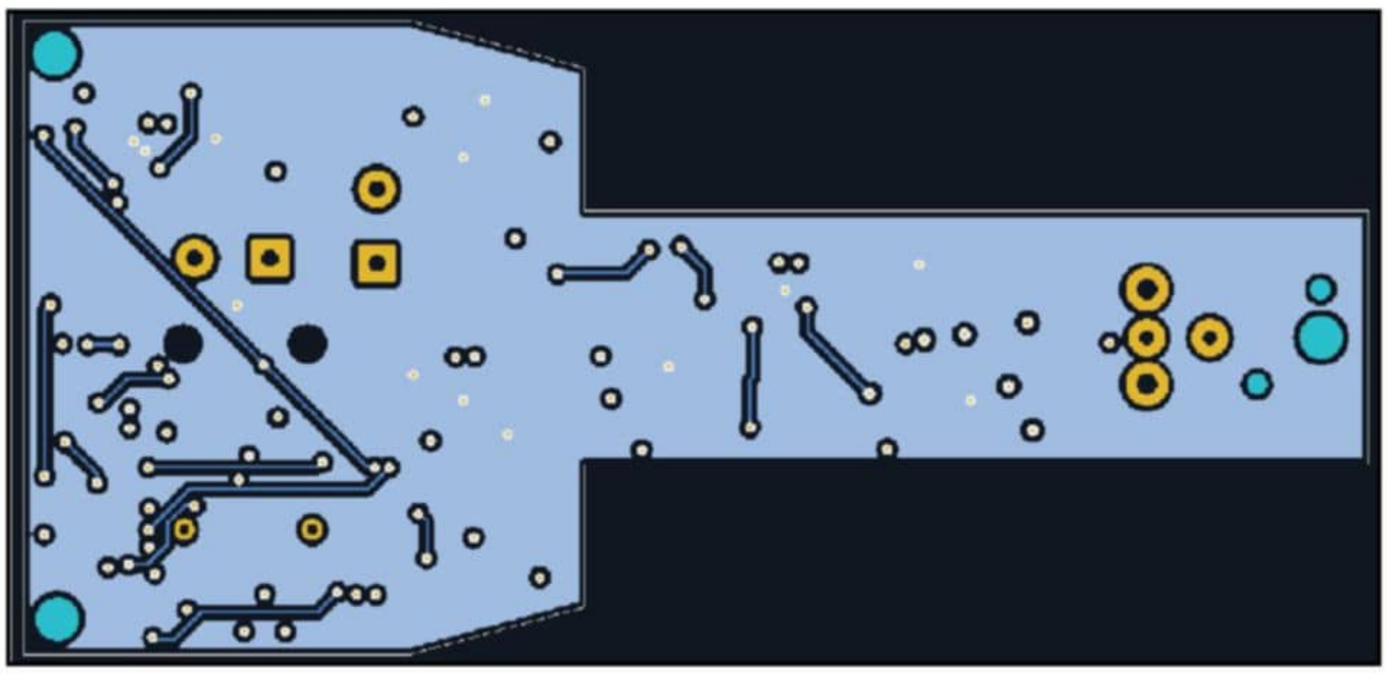


Figure 9: The bottom layer of this board is filled solid to carry the V/2 half-supply around the board, plus a few traces that couldn’t go on top.

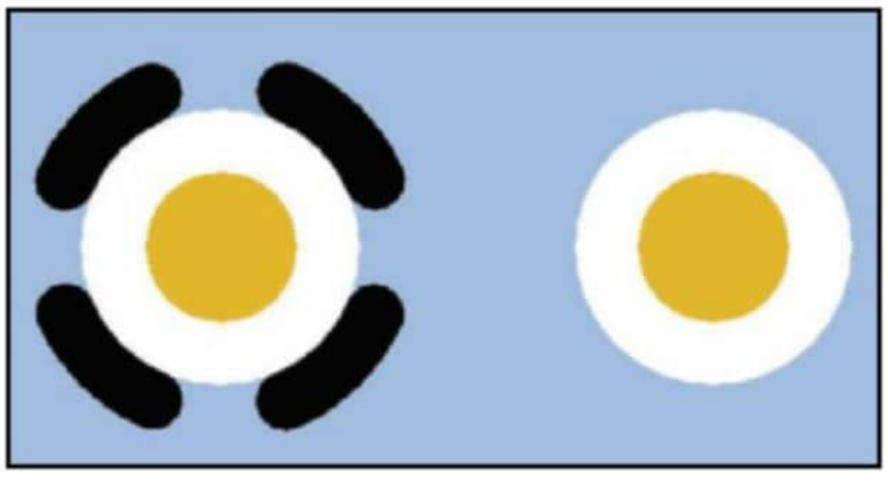


Figure 10: When a pad is used to connect a through-hole component to a filled area, the thermal relief (at left) prevents the fill from acting as a heatsink when soldering.

circuits high voltage isn't an issue, but manufacturing tolerances are. Many PCB manufacturers advise traces no narrower and no closer than 0.008", though tighter tolerances might be available at a higher cost per board. If you stick with traces and spacing no smaller than 0.01" you're guaranteed to be safe. This is another way KiCad is superior to ExpressPCB—KiCad won't even let you put traces closer than the minimum specified in Preferences. Nor will it let you make an improper connection between components.

Never use 90-degree angles for traces. Sharp angles can end up narrower than the expected trace width when the copper is etched, and they are also more likely to peel off the board. It's best to aim for 45 degrees, and here again KiCad is superior with an option to enforce that angle.

Short runs are always better than going around the block, even if that means using a via to get to the bottom layer. It might seem like cheating if you need to use a via, but electrically it's better to keep runs short and close to the connected components. **Figure 11** shows the top and bottom layers at the same time, with two ways to connect R83 to R84. Using a via is clearly preferred because it's much shorter. Notice that the "No" trace passes between the terminals of C29 and C30. That by itself is fine, and there's nothing wrong with routing a trace under most components between the pads if it fits.

ExpressPCB includes footprints for hundreds of standard parts, both through-hole and SMD, and KiCad includes more than 12,000 with more being contributed by users all the time. But it's inevitable that you'll eventually need to modify a part's footprint, or design new footprints from scratch. Creating and editing footprints is easier in ExpressPCB than in KiCad, though many of the stock SMD outlines in ExpressPCB are slightly larger than needed. So when trying to pack as many parts as possible onto a small board, you'll probably end up making some of them smaller.

Among its many View options, KiCad can show "courtyards"—the thin purple rectangles shown in Figure 8 surrounding each part. These display the physical size of the part as defined in its footprint. This lets you ensure that adjacent parts have enough clearance to avoid bumping into each other.

For solid-state devices having more than two pins, it's important to verify that the pin numbers on your parts match the pin numbers on the board. Most transistors assign Pin 1 to the Base, Pin 2 to the Emitter, and Pin 3 to the Collector. But some are different. And some two-wire parts, such as Zener diodes, come in three-pin packages with one pin unused. The dual transistors I used in this project are available with two different pin-outs, so be sure you know which type you have!

It's best to mount pots, switches, and connectors directly on the PC board to avoid hand wiring when possible. But sometimes these parts need to be on a separate panel. In that case you'll add pads with through-holes to accept the connecting wires. Figure 8 shows two such pairs of pads for potentiometer P1 and switch SW1. Because these parts are in the audio signal path, I'm using shielded wire with the shield connected to the lower impedance. One is the output of an op-amp and the other is ground, with square pads identifying the shield connections.

ExpressPCB and KiCad both have plenty of standard pads with holes to choose from, but you need to choose pads with holes large enough for the wire's gauge. Since most wires are stranded, you'll probably tin the ends before pushing them through the pad holes. So figure the pad holes should be 15% to 20% larger than the wire diameter. There's a terrific page on Wikipedia that lists the diameter for all of the standard wire gauges, plus other useful info, linked under Resources. Or better, measure a sample piece of wire yourself with a caliper after tinning it, as I do.

If you're making a large number of very small PC boards, you might consider panelizing them into a larger group of connected "snap-apart" boards. If your board is, say, 1" by 2", you could group six or 10 of them together with ¼" between. The board maker will saw partway through the board on each side, making it easy to snap them apart with your hands. Panelizing might make the boards easier for you to handle during assembly, or reduce the cost of each board. Just as important, if an outside shop does the assembly for you, they may require a certain minimum board size their automated machinery can handle. Many board makers can panelize boards for you. Just send them the files for one board, and they'll figure out how to group and score them.

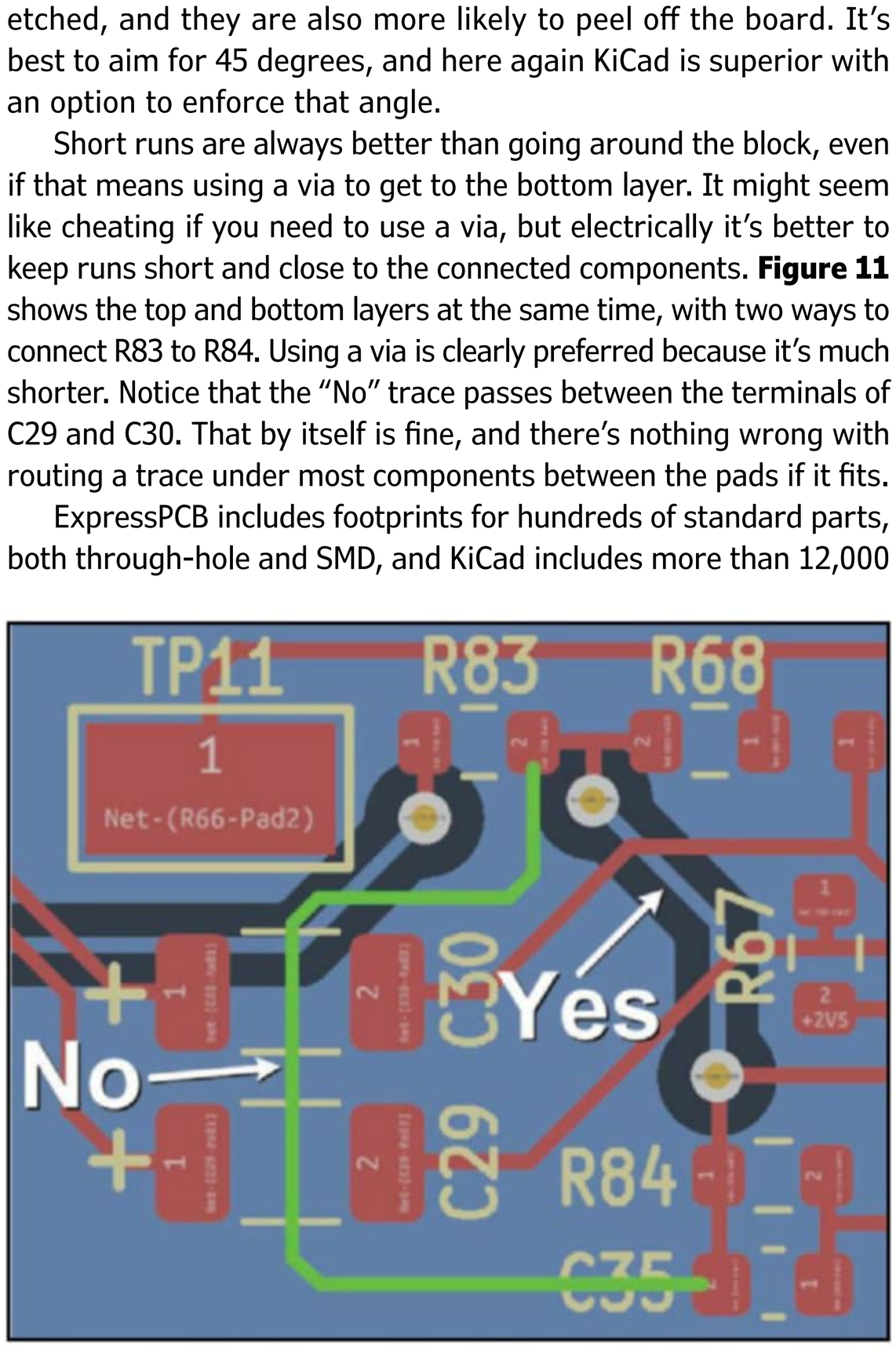


Figure 11: Short runs are always better than long runs, even when vias are needed to get to a trace on the bottom layer.

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 50 years. His Cello Rondo music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press, now in its second edition, is available at www.amazon.com and on Ethan's own website www.ethanwiner.com. Ethan is also a principal at RealTraps, a US manufacturer of high-quality acoustic treatment.

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and audio coupling, ceramic for IC power supply bypass, and film when low distortion is needed for audio and other linear signals passing through the cap. My previous article, "Stop Worrying About Coupling Capacitors!" (*audioXpress*, May 2020) shows distortion amounts measured for a variety of capacitor types.

Ceramic caps are used for power supply bypass with ICs because they're inexpensive, small, and their series inductance is very low. But there are several things to consider when choosing ceramic capacitors, especially the smallest SMD types. As mentioned in my earlier Null Tester article (*audioXpress*, June 2019), the value of some ceramic capacitor types is reduced dramatically as the DC voltage across them rises. This is due to their voltage coefficient. So a 10µF filter cap rated at 10V might indeed be 10µF with 1V DC across it, but only 5µF—half as much—when used to bypass a 6V power supply. Therefore, you often need to buy parts having more capacitance than required, based on the percentage reduction shown in the cap's spec sheet under DC Bias Characteristics or similar wording. Class 2 ceramic capacitors (X5R, X7R) are pretty bad in this regard; Class 3 (Y5V, Z5U) are even worse, and their capacitance further varies with temperature.

An excellent article with more detail and explanations about ceramic capacitors is on the All About Circuits website (see Resources). Another helpful article is on the Murata website. **Figure 12** shows the typical capacitance change for Class 2 and Class 3 capacitors versus DC bias voltage, derived from data on the Murata website. C0G and NP0 ceramic capacitors have a Class 1 dielectric, which is much better, and so are often good enough to send audio through when a film cap won't fit. C0G/NP0 caps are larger than X5R and Y5V types, though not nearly as large as film caps having the same capacitance and maximum voltage ratings. A much more comprehensive explanation of ceramic capacitor properties and selection criteria can be found on Wikipedia (see Resources).

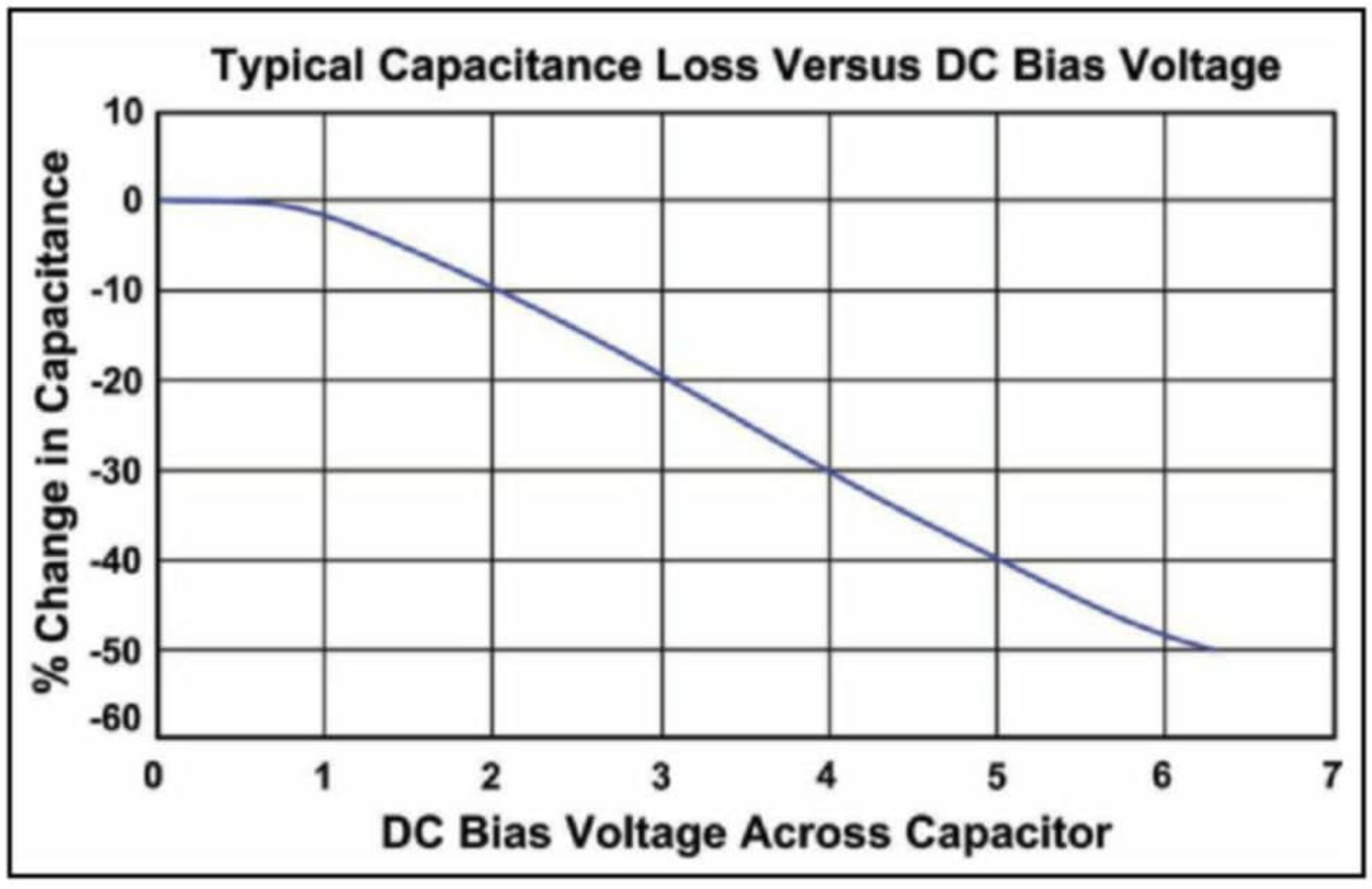

Figure 12: With Class 2 and Class 3 ceramic capacitors, their capacitance reduces as the applied DC bias voltage increases.

The capacitance change of Class 2 and Class 3 capacitors due to applied voltage creates distortion when audio and other linear signals only partially pass through them. These capacitors are also microphonic due to their piezoelectric properties. When only the smallest capacitor type will fit, my expert engineer friend Bill Eppler told me that he solders these caps onto the board sideways rather than with the wider side face down as usual. This reduces flexing of the capacitor's internal leaves when the circuit board bends due to vibration. If a ceramic cap doesn't flex, it will not create piezoelectric noise.

When laying out your PC board, you should place bypass caps as close as possible to their associated IC, with their traces and vias to the ground and power layers as short as you can manage. **Figure 13** shows a close-up of a logic chip and its bypass cap CB6. The trace between A5 pin 4 and its ground via above is only 1/8th of an inch long, so it's a lot closer than it looks! If this circuit needed to operate at much higher frequencies, it would not be wrong to add a second via to the ground plane right next to Pin 4.

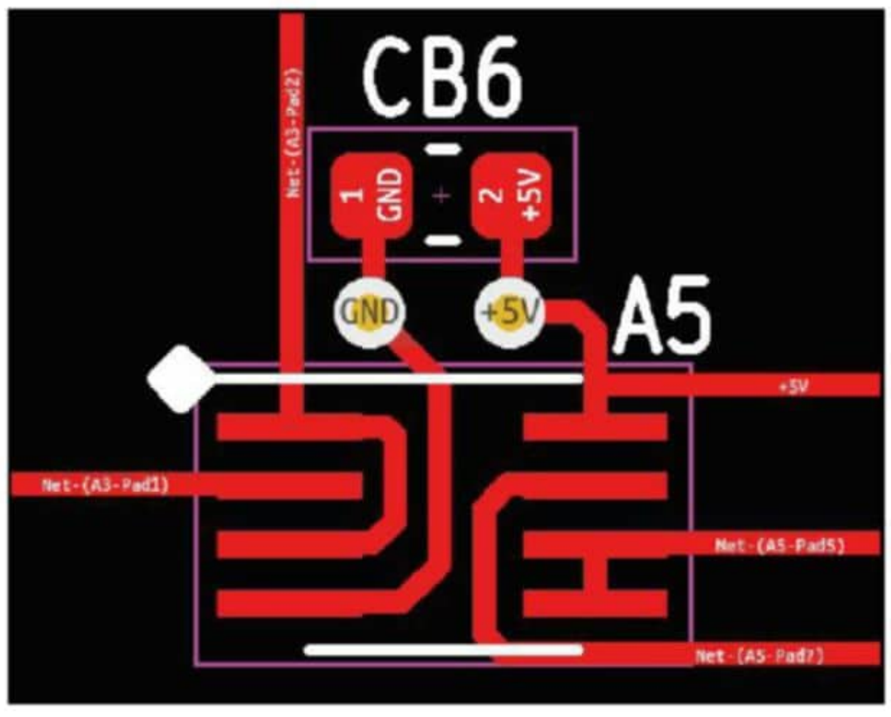

Figure 13: When designing your board, keep bypass caps as close as possible to the IC they're protecting.

And Finally

The last step is to order the board, and optionally a solder mask template, from your chosen manufacturer. ExpressPCB is more expensive than ordering boards from China, even with international air shipping. But the software is not difficult to learn and use, and ordering is very easy right from within

the PC board design program. You answer a few simple questions, click Get Price, and a few seconds later you know the final cost including shipping. So I gladly stuck with ExpressPCB until I needed ENIG gold plating, which it does not offer. My time and effort are worth more than the price difference between board makers.

KiCad is not tied to any particular vendor, so it outputs standard Gerber files that any manufacturer can use. However, it's much more complicated to learn and use than ExpressPCB. If I had to start over with what I know now I would choose ExpressPCB again. But beware, the error checking in ExpressPCB is not as thorough as in KiCad. I designed my current board initially in ExpressPCB, and accidentally shorted the Ground and V/2 layers with vias rendering the board useless. KiCad automatically prevents such errors, so that could never happen.

Getting quotes and ordering boards from major companies can also be tricky. I found the AllPCB website to be rather confusing compared to ExpressPCB because there are so many options, such as copper thickness, board material, and choice of silk screen color. Of course, professional engineers will appreciate all those options! Thankfully, AllPCB has an online chat option where you can ask questions. Even easier, you can click a button to send them your files with a written description, and one of their salespeople will reply by email with a quote. ax

Resources

AllPCB | www.allpcb.com

"Ceramic Capacitor FAQs," Murata, www.murata.com/en-us/support/faqs/capacitor/ceramiccapacitor/char/0005

ExpressPCB | www.expresspcb.com

KiCad | www.kicad.org

R. Keim, "X7R, X5R, C0G…: A Concise Guide to Ceramic Capacitor Types," March 30, 2018, www.allaboutcircuits.com/technical-articles/x7r-x5r-c0g...-a-concise-guide-to-ceramic-capacitor-types

Wikipedia, "American wire gauge," https://en.wikipedia.org/wiki/American_wire_gauge

Wikepedia, "Ceramic capacitors," https://en.wikipedia.org/wiki/Ceramic_capacitor

E. Winer, "You Can DIY! Building a Null Tester Device," *audioXpress*, June 2019, https://audioxpress.com/article/you-can-diy-building-a-null-tester-device

E. Winer, "Stop Worrying About Coupling Capacitors!" *audioXpress*, May 2020, https://audioxpress.com/article/practical-test-measurement-stop-worrying-about-coupling-capacitors

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Power Transformer Parameters, Selection, and Testing

Part 5 — Losses

The initial group of articles focused on the history of transformer cores, construction methods and materials, and various testing methods. After having discussed the parameters of transformers, this article expands on their losses.

By
Chuck Hansen

When a load, R_L, is connected to the secondary winding, the secondary current (Is) creates a magnetic field back-EMF that opposes the primary flux (Lenz's law) shown as E_{back} in Figure 7 (see *audioXpress*, February 2023). The primary current then increases by the amount necessary to restore the original flux density supported by the primary voltage. Because of the large primary and secondary winding inductances, the rate of current change depends on the transformer R-L time constants. As long as the steady-state condition exists, the average flux density will remain the same.

If the load current increases, the flux density in the core decreases due to the higher voltage drop across the primary winding resistance RP. The secondary impedance Z_s, composed of resistance R_s and inductance X_s, causes a further reduction in the secondary voltage V_s. The ratio of the difference between the no-load secondary voltage (V_{SNL}) and the rated-load secondary voltage (V_{SRL}) for a fixed primary voltage is called the regulation factor. Sometimes the specified regulation factor is based on half-load, so always read the data sheet details.

Regulation Factor (ε) = $(V_{SNL} - V_{SRL})/V_{SRL}$

Smaller transformers tend to have worse regulation factors. The thickness of the insulation is limited by the hi-pot voltage requirements, and the thickness of the wire insulation doesn't vary much with wire gauge, so there is less copper area with smaller gauges. The effective copper area for the space inside the core window available for the windings is also reduced.

When regulation factor is specified by the manufacturer, it is usually with a resistive load. It will be worse for inductive loads and might be better for capacitive loads, which could cancel Xs. In tests I ran on various sizes of E-I lamination transformers with a resistive load bank, the regulation factor varied between 0.05 (5%) and 0.20 (20%). These load resistors do have some amount of inductance, I did not use non-inductive types such as Ayrton-Perry or bifilar-wound resistors. They are all metal-case RE or HS types from 25W to 250W, Mil RW types from 3 to 10 W, or rectangular cement resistors from 5W to 10W.

If you increase the primary voltage, or decrease the secondary load impedance with a fixed primary voltage, you start to enter the horizontal saturation portion of the B-H curve where no further flux B can be generated regardless of the amount of primary current used (**Figure 12**).

Once the B flux density passes the peak permeability μp of the transformer core, (the Q point on the blue line) the relative permeability μr (green

line) starts to decrease at a rate determined by μD, the differential permeability. More magnetizing current H is required to generate the B-H curve. Eventually the core permeability reaches unity μ0 when it becomes fully saturated.

Operating at high flux densities also increases the third harmonic as well as other higher odd-order harmonics. At this point only the primary winding resistance limits the current. As the primary voltage decreases, the B-H curve traces the path shown back in Figure 8 (see *audioXpress*, February 2023).

Aerospace AC power generators usually have to supply 150% rated current for two to five minutes (depending on the cooling method), 200% for five seconds to allow for large motor starting, and up to 300% rated current (but not rated voltage) for five seconds to allow time to clear short circuit faults. Three-phase AC generators also have unbalanced phase load requirements of up to 30%. These generators, and any transformers powered by these generators, operate at a lower full-load flux density that allows for a higher permeability and higher efficiency. Q is the point on the B-H curve for peak permeability. SWaP-c has its limitations, but certain specification requirements can have their advantages for EM design.

Large utility generators, on the other hand, have limited overload capability. The prime mover (nuclear steam power, fossil fuel steam power, hydro power) is designed to operate between 70% load and rated load, commensurate with the cost of generation. An overload could cause the grid frequency to drop, requiring load shedding or the addition of fast-response gas turbines or large diesel peaking units to be added to the grid to maintain precise frequency control and system stability.

The increase in plug-in hybrid electric vehicles (PHEV) and battery electric vehicles (BEV) raises issues of their increasing electrical load on the utility power grid. Large numbers of these vehicles needing charging at the same time could potentially destabilize the grid. A smart electric vehicle charge controller has been developed that can be installed in a home charging unit, at businesses or at public access charging stations. It continuously monitors the power grid frequency and automatically reduces the vehicle charging load and helps maintain system frequency [2]. They can respond much faster than the high inertia of large utility generators.

Dielectric Losses in the Insulation System

There is parasitic capacitance in a transformer from winding to winding, and from windings to the steel transformer core. The amount of total parasitic current flow depends on the type and thickness of the dielectric used on the copper wires and the inter-winding insulation, and the spacing between the wires and core.

The capacitive current flow increases with frequency and produces waste heat that decreases the efficiency. We use low-loss, low dielectric constant materials in aerospace wherever possible. PTFE Teflon wire insulation has a lower dielectric constant than irradiated PVC insulation, 2.02 vs. 2.7.

Copper Losses

Current density is the current per unit area of wire in amps per square inch or amps per square mm (A/in^2 or A/mm^2), depending on the core catalog used. Current density is a design trade-off between size, weight, and temperature. Lower current density means that the transformer is larger, heavier and costs more, but more efficient than one operated at a higher current density.

From what I have compiled from online data, commercial transformers have wire current densities ranging from 2 A/mm^2 to 5 A/mm^2, with the smaller gauges designed for higher current densities for the reasons I previously mentioned under Regulation Factor; the minimum thickness of the insulation is limited by the hi-pot voltage requirements, and the wire insulation thickness doesn't change very much with wire gauge.

Copper losses in transformers, rotating machines and other electromagnetic devices are caused by the I^2R wattage in the copper windings. Higher frequencies increase the skin-effect losses, which is the tendency of high-frequency alternating currents to crowd toward the surface of the windings, increasing the current density. As the devices heat up the resistance of the copper increases 39% per 100°C, and so do the copper losses. The only metal with a lower resistivity than copper is silver.

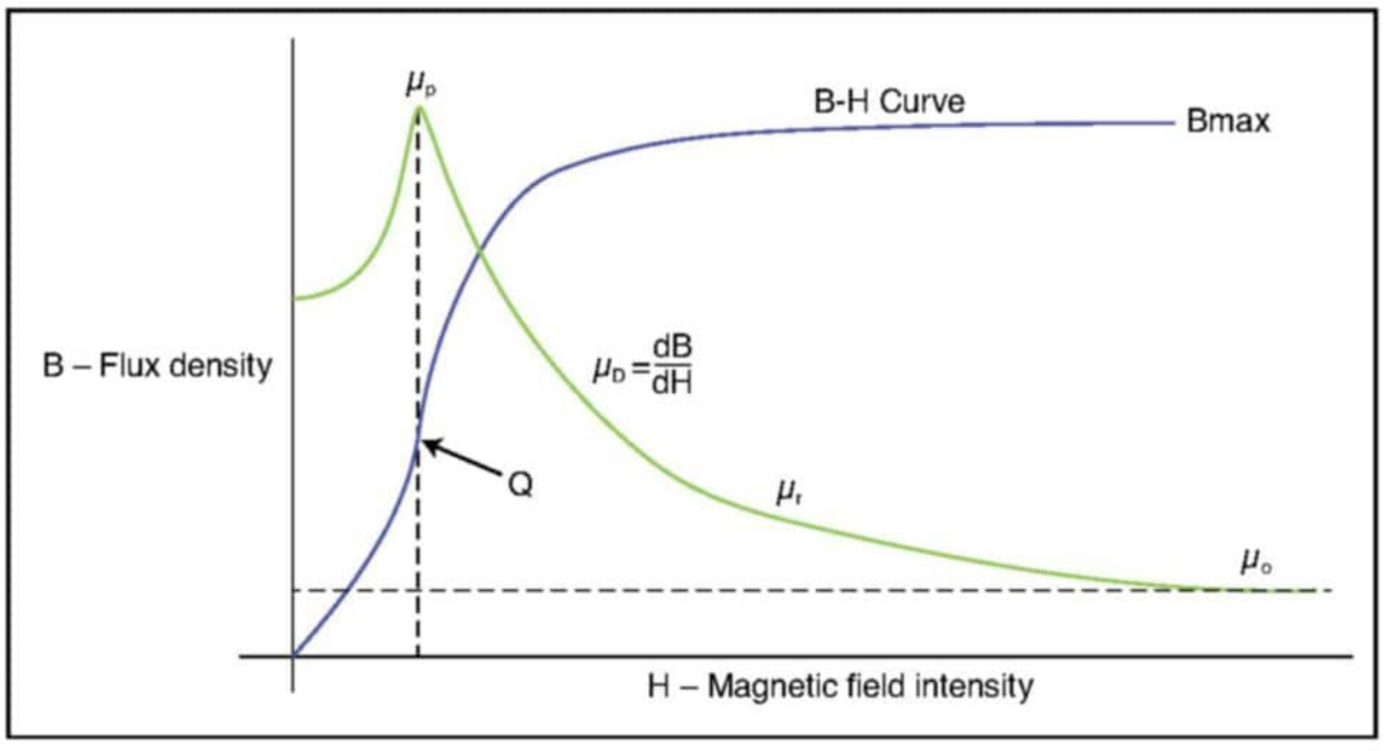

Figure 12: Differential permeability vs. magnetic field intensity (Image Source: COMSOL)

Some large utility transformers use sheet aluminum for the low voltage, high-current secondary windings to save cost and sometimes weight. The conductor area has to increase to keep the winding resistance the same as the equivalent copper winding. The resistivity of aluminum is 2.65 × 10^{-8}Ω.cm, while copper is 1.68 × 10^{-8}Ω.cm, so the area of the aluminum has to increase by 1.58 times.

Aluminum is also used for high-voltage transmission lines, to save weight and reduce sag at high temperatures. The density of copper is 9.0 g/cc while aluminum is 2.7 g/cc, or just 30% for the same volume. Adjusted for the larger area of aluminum needed for the same resistance, an aluminum transmission line is 47% of the weight of copper for the same current rating. Copper and aluminum wires are soft materials, so they are wound around a steel core to provide the tensile strength needed to support the lines over the long distances from tower to tower.

For E-I cores using symmetrical bobbins and split windings, there is an advantage in keeping the primary and secondary winding copper losses equal for cooling purposes. The line-voltage primary has more turns of thinner gauge wire than a step-down secondary winding with fewer turns of heavier gauge wire.

For a 120V AC transformer with a full-load primary current of 1A, the transformer requires a primary wire gauge of AWG 24, assuming an ampacity of 400 circular mils per amp for Formvar insulated round magnet wire, or the ML insulated round wire we use in the smaller self-cooled transformers we make for aerospace applications. If the full-load secondary current is 2A, which would require a wire gauge of AWG 21.

Note that every three numerical steps down in wire gauge have essentially twice the wire cross-sectional area and twice the ampacity. The area of AWG 24 is 404 circular mils, while the area of AWG 21 is 810 circular mils. The next three numerical steps down to 18 gauge has an area of 1624 circular mils, and so forth. Insulated round 24-gauge magnet wire allows for 42 linear turns-per-inch (TPI), while AWG 21 wire only allows 30 linear TPI.

For C and E cores with square or rectangular core boxes as shown in Photo 12 (see *audioXpress*, January 2023), we usually do layer winding with the primary winding applied first. This keeps the overall length of the wire shorter than if the primary was wound over the secondary, and minimizes the primary winding copper loss. This is important because, if the primary is buried under a layer of insulation and one or more secondary windings, it will not dissipate heat as well as the outer winding(s). Plastic bobbins allow for side-by-side windings, which also reduces the inter-winding capacitance.

The wire gauges and coil locations for higher power transformers are decided on the basis of a thermal analysis using a program such as SolidWorks, which calculates heat transfer by conduction, convection and radiation, as well as thermally induced stress.

The winding machines for C and E cores work like a fishing pole reel with the core box or bobbin as the spool. The operator adjusts the wire feeder based on the wire gauge being wound and the width of the winding layer.

The wire feeder moves back and forth, winding the wire in even layers on the core box. The wire cannot be wound all the way out to the ends of a core box since it could slip off the end. A small gap is left between the ends of the winding and the ends of the core box to prevent that. Bobbins have flanges that hold the wires in place.

With a core box, a layer of thin Kapton or reinforced fiberglass tape is applied over the primary winding, then the secondary winding is wound over the top of the primary. If there is more than one secondary each one is wound on top of the other, with insulating tape in between windings, and then over the top of the outside winding. The insulating tape prevents hi-pot voltage failure between the windings. The core box prevents hi-pot voltage failure from the primary winding to the transformer core. If there are low voltage secondaries with few turns, they may be wound

Photo 16: Generator stator skew to minimize slot harmonics (Image Source: The Back Shed)

side by side. Bobbins solve a lot of these problems and minimize the amount of tape needed. Toroidal cores, O cores and R Cores require special winding machines, which will be covered in detail later in this article series.

Skin-Effect Losses

When a DC current passes through a conductor, the current density is uniform across the entire cross-section of the conductor. With an AC current, as the frequency increases there is an increase in the current density toward the surface of the conductor. The effective conductor resistance increases due to this non-uniform current density. Skin-effect also increases with the diameter of the conductor. Even at 50Hz there is a small increase in the current density near the conductor surface as compared to DC current.

Litz (Litzendraht) wire can help reduce the skin-effect of high-frequency AC windings, usually from 10kHz to 1MHz. It is made up of many fine conductors, insulated from each other, and interwoven such that every conductor equally occupies every location in the total cross-section of the wire bundle. This ensures that the current in every strand is equal. We have used Litz wire for high-power 400Hz transformers, and have even made generator stators windings with Litz wires for some higher frequency applications.

Skin-effect is also a factor in core losses because the iron, nickel or cobalt cores are conductors, and the eddy current wants to flow across the laminations (see Figure 10 in *audioXpress*, February 2023).

Losses Due to Power Line Harmonics

Harmonic distortion in the 50Hz or 60Hz AC voltage produces increased losses in transformers as well as motors and other electromagnetic devices. In order to see another of the ways in which harmonics occur, it will be necessary to take a brief side-trip into the design of AC generators.

The production of harmonics in the generator AC voltage has several variables. Since the steady-state AC waveform has half-wave symmetry about zero, there are no steady-state even harmonics produced, although load transients can produce a short-duration DC offset in the ac waveform. The arrangement of the coil windings in the stator slots (called the winding pitch and phase belt) is used to cancel the third harmonic and all its multiples (called triplens) in aerospace generators, as well as smaller single-phase 50Hz/60Hz generators that are connected directly to low voltage AC loads, or those used for backup generators in case of loss of AC power in your home.

For large four-wire, grounded-wye (Y) utility generators the pitch and phase belt are arranged to allow the third harmonic, which is then canceled in the high-voltage step-up transformer secondary winding. They are connected in three-phase delta (Δ) instead of the three-phase four-wire Y connection from the generator to the transformer primary. The generator Y neutral is connected to ground through a large neutral reactor (inductor), which limits unbalanced phase currents or overloads. The turbines or engines that power these large generators have a limited capability to support overloads.

The major odd harmonics produced in the air gap by an optimum fractional winding pitch are the 5th, 7th, 11th, and 13th. Higher frequency harmonics are caused by the area discontinuity of the stator slots that hold the phase conductors. They are canceled by skewing the angle of the slots by one slot width across the magnetic length of the stator core (**Photo 16**).

The harmonic current and voltage cause increased heating because of the increase in iron and copper losses, since they are both frequency dependent. Copper losses are further increased due to the skin effect caused by the higher frequency harmonics. All this reduces the efficiency of the generator, as well as all the motors, solenoids, inductors and transformers connected to the ac power.

k-factor transformers are designed to safely withstand higher losses from harmonic effects by increasing the core area, using thinner core tapes and/or using multi-winding conductors or Litz wire to limit skin effect.

The large-scale increase in solar and wind power potentially introduces high-frequency harmonics. Solar panels generate DC voltage. The efficiency of each panel is maximized by a solid-state device called a maximum power-point tracking (MPPT) controller, a DC-to-DC converter that optimizes the impedance match of each solar panels to a battery storage bank. The battery power is converted to the proper AC voltage and frequency

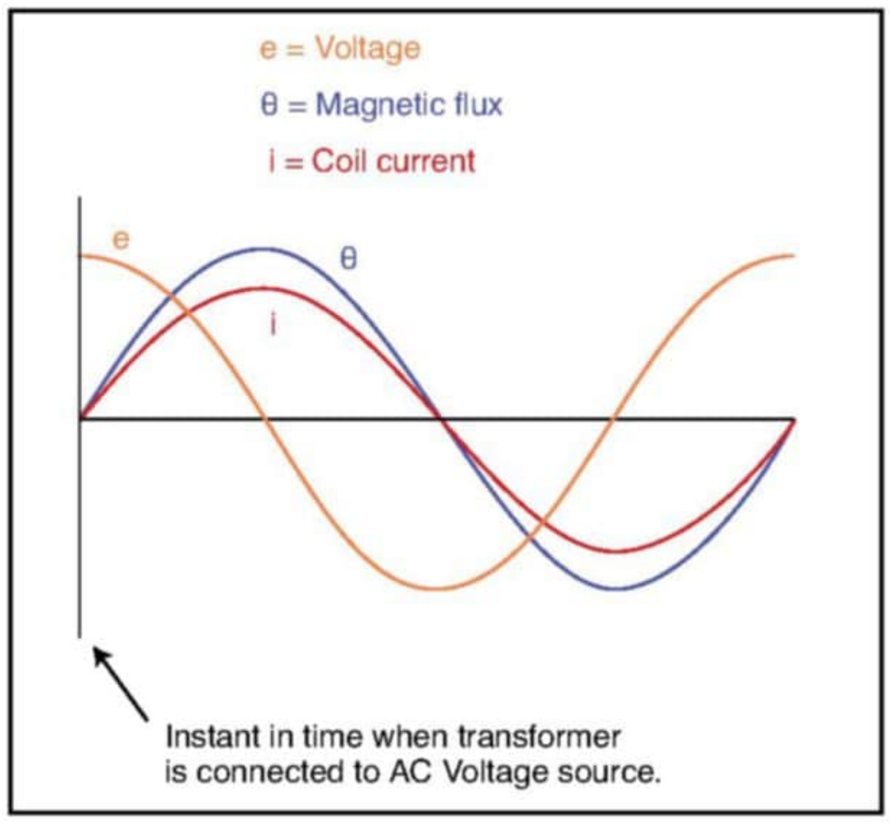

Figure 13: Transformer current when the AC voltage reaches its peak (Image Source: All About Circuits)

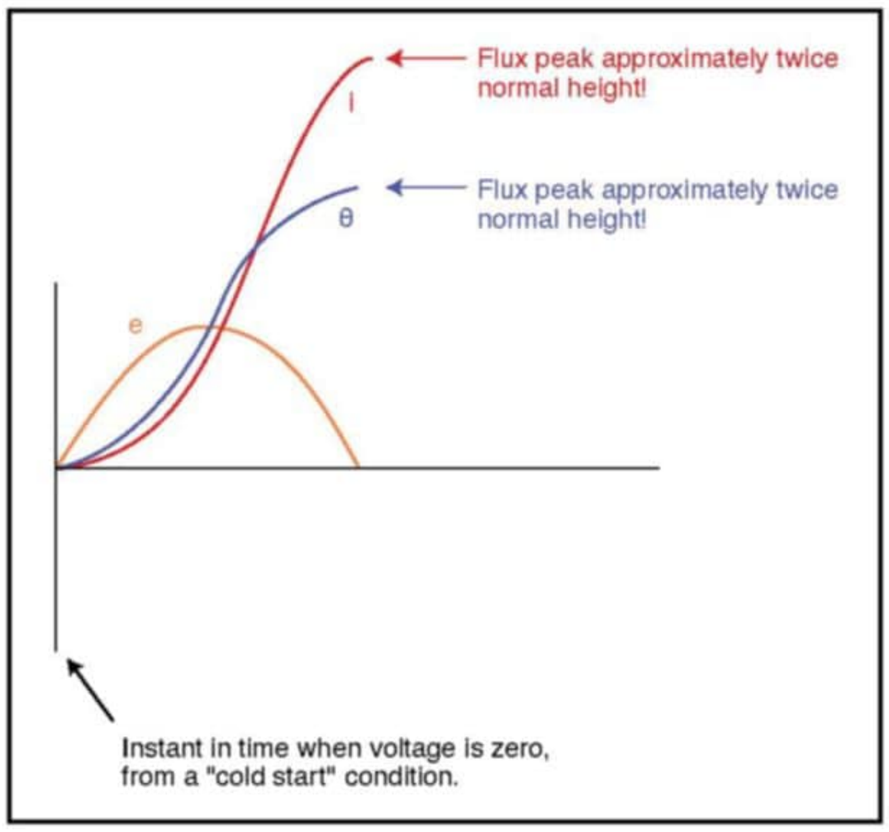

Figure 14: Transformer current when the AC voltage crosses zero (Image Source: All About Circuits)

References

[2] "Smart Frequency-Sensing Charge Controller for Electric Vehicles," Argonne National Laboratories, Lemont IL; www.anl.gov, US Patent 8,789,249 B2, August 5, 2011, Inventor: Vladimir Koritarov.

[3] E. Csanyi, "Testing of Power Transformer — Measurement of Zero-Sequence Impedance," Electrical Engineering Portal (EEP), July 2014, https://electrical-engineering-portal.com/testing-of-power-transformer-measurement-of-zero-sequence-impedance

[4] "Practical Considerations—Transformers," Chapter 9, Transformer Primary Inrush Current, All About Circuits, www.allaboutcircuits.com/textbook/alternating-current/chpt-9/practical-considerations-transformers

Resources

L. Liu, "How the B-H Curve Affects a Magnetic Analysis (and How to Improve It), COMSOL Blog, November 2019, www.comsol.com/blogs/how-the-b-h-curve-affects-a-magnetic-analysis-and-how-to-improve-it

Practical Considerations—Transformers," All About Circuits, www.allaboutcircuits.com/textbook/alternating-current/chpt-9/practical-considerations-transformers

The Back Shed, https://thebackshed.com/forum

for the intended local load or to utility power. Some MPPT controllers generate large amounts of electromagnetic interference (EMI) in their AC output due to their pulse-width modulation (PWM) switching, as reported by the Amateur Radio Relay League (ARRL) some time ago.

There may also be a panel tracking controller that adjusts the angle and elevation of the panel so it is perpendicular to the sun as long as possible.

Modern horizontal-axis wind turbines are attached to usually three long slow-turning blades whose pitch is constantly adjusted for the optimum speed. An AC generator is contained in the housing behind the blades. There is also a wind angle tracking device that keeps the blades facing into the prevailing wind.

Either a direct-drive rare-earth magnet generator or a synchronous AC generator with a speed-increaser gearbox, is coupled to the blade hub. The electronics that control blade pitch and yaw are housed in the nacelle that contains the generator and controls. These are also PWM controllers that can generate EMI. Sometimes a power converter is contained in the nacelle as well.

Inrush Current Transients

Another issue occurs when large utility transformers are first energized. The transformer low-voltage (13.8kV) primary inrush current could theoretically be up to 12 times the rated load current, but the generator neutral reactor again limits the transient inrush current and any current flow due to unbalanced loading.

This high inrush current pushes the transformer towards saturation, producing third harmonics and possibly a short-time DC component called a zero sequence, until the inrush resolves itself [3]. In order to prevent false tripping of the generator and/or transformer differential protection relays, these relays have special third harmonic restrain circuits.

A similar situation occurs on a much smaller scale when a toroidal, O core or R core transformer is energized from household 120V AC/240V AC lines, especially with a capacitive secondary load. If the core saturates (B_{max}), the winding inductance appears greatly reduced, and only the resistance of the primary windings and the very low impedance of the AC line limit the current. This is why the time delay fuse current ratings for these transformers is usually larger than for E-I core transformers.

There is normally a 90° phase lag between current and voltage in a transformer due to the large primary inductance. If the AC voltage is connected to the transformer primary just as the AC voltage waveform reaches its peak, the

primary inrush current is symmetrical about the zero axis (**Figure 13**) and the coil current will be at its minimum [4].

If the voltage is connected just as the primary AC voltage waveform crosses zero, the half-cycle saturation produces an initial asymmetrical DC-offset, and the initial flux peak will be twice the steady-state value. Harmonic-rich waveforms can be generated that can affect the load (**Figure 14**) [4]. With a really low transformer primary winding resistance, the sub-cycle transient current could reach up to 10 times rated current, depending on the type of transformer core material, the winding resistance and the source impedance of the ac voltage. The red lines shown in both Figure 13 and Figure 14 are the coil currents.

With a higher remanence core material, some of the dipoles in the core could be left in one direction during the sudden application of zero-crossing voltage and resulting peak primary current. It could take a number of ac cycles to "erase" the dipole memory from the core and restore a symmetrical current waveform (one with no DC offset). Toroidal, O and R core transformers are especially prone to this core magnetization since they don't have air gaps. Just a small amount of DC offset in the AC waveform can cause a big increase in their primary current.

Next Month

Next month we will look at commercial AC power quality both in North America and abroad, electrical component reliability, and the design of current transformers. ax

About the Author

Chuck Hansen is an Electrical Engineer and holds five patents in his field of electric power engineering. He has worked in both the electric utility and aerospace electric power industries. He was assigned as the Instrument and Controls Startup Group Lead Engineer for two nuclear power generating units. He was the Electrical Systems and Controls Supervisor for a number of military programs as well as commercial and business jet electric power programs. He has written two books for Audio Amateur publications, and more than 260 magazine articles. He began building vacuum-tube audio equipment in college, and enjoys restoring and modifying audio equipment and test equipment. In his most recent roles as Senior Design Engineer and Lab Calibration Manager, he designed specialized test equipment particular to the aerospace electric power sector.

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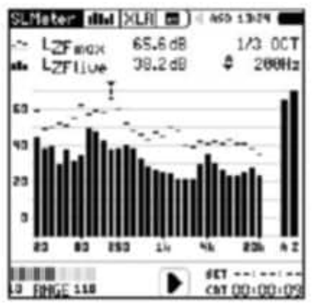

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Hollow-State Electronics

Accuracy in LTspice Modeling

Sometimes we need to consider the human factor when using circuit design simulation tools. In this article, our hollow-state expert details an error that occurred in a past article, which is now being corrected.

By
Richard Honeycutt

While I was in undergraduate school in the late 1960s, Hewlett Packard introduced its first minicomputer, which, if memory serves, had 8kB of RAM (twice that of the IBM 1620 the college used to teach programming), and a paper tape punch/reader for programming and output. The physics department bought one, and I programmed software to calculate the resonant modes of a rectangular room as part of my senior thesis project. A few years later, when I began reading scientific papers reporting the results of computer modeling, it did not yet dawn on me that what I had done on that project was in fact computer modeling. Consequently, I was hesitant at first to trust the conclusions drawn from computer modeling of physical systems.

At the time, I was blissfully unaware of the development of the Electrical Circuit Analysis Program (ECAP) by IBM in the 1960s, or the Simulation Program with Integrated Circuit Emphasis (SPICE) in 1973. The descendants of SPICE written for personal computers (PSPICE, LTSPICE, and several others) came along in the early to mid-1980s. It was then that I became familiar with these circuit design tools as part of my duties as a college electronics professor. I found that while care was necessary in building circuit models to run on these simulation programs, they produced excellent results.

Some common error sources included:

- Poor component models, especially for transformers and vacuum tubes
- Parasitics such as stray capacitance, which were especially problematic in high-frequency analysis.
- Human error: undoubtedly the most prodigious error source.

Software companies provided component models for ideal resistors, capacitors, inductors, transistors, solid-state diodes, and later, ICs. Users had to (and for the most part, still have to) build their own transformer models. Doing so required knowing—at a minimum—the primary and secondary inductances and the coefficient of coupling. For more accurate models, the winding resistances, leakage inductance, and interwinding capacitances must be known. The values of these parameters are often hard to find.

Since most engineers assumed that vacuum tubes were a dead end, no hollow-state models were available at first. However, the circuit modeling programs included the capability to accommodate reasonably accurate vacuum tube models, and a number of dedicated hobbyists began creating vacuum tube models and offering them free in the Internet.

Today, good LTspice vacuum tube models are available for almost any tube that is still in production and also for many obsolete tubes. Unfortunately, some OEM tubes such as several Western Electric types do not have available models, mainly because only “pre-owned” tubes of these types can be purchased today. In these cases, hobbyists often find a similar tube for which a

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model is available, and substitute that one in their circuit models for simulation.

Why Bother with Modeling?

Although today, circuit modeling is ubiquitous in solid-state engineering, it is less common in the hollow-state world. I know several excellent engineers who design and build hollow-state equipment as a hobby, but I am not personally acquainted with any professional hollow-state

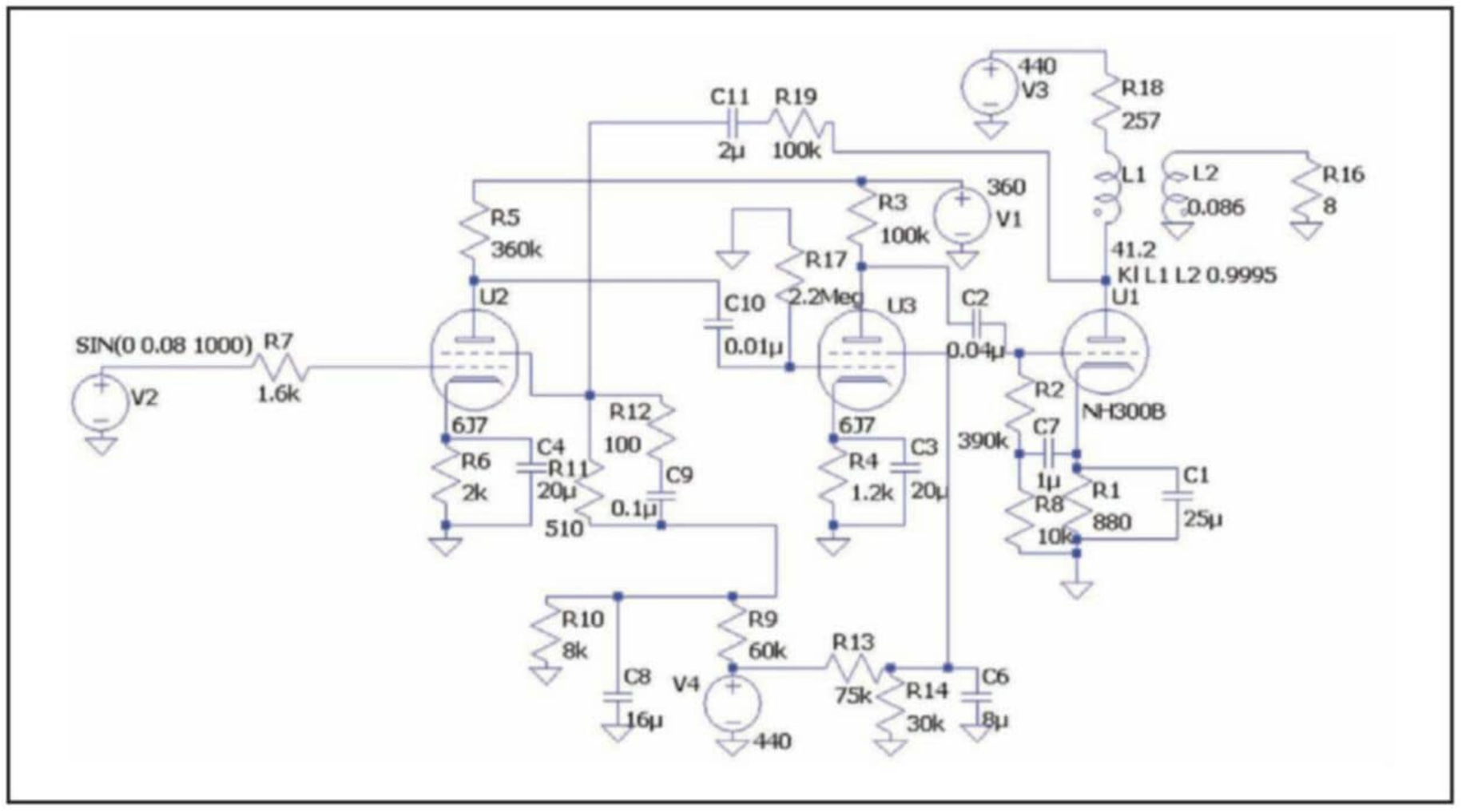

Figure 1: This model for the Western Electric (WE) 91A includes the second-stage grid-leak resistor R17 as well as a few other corrections.

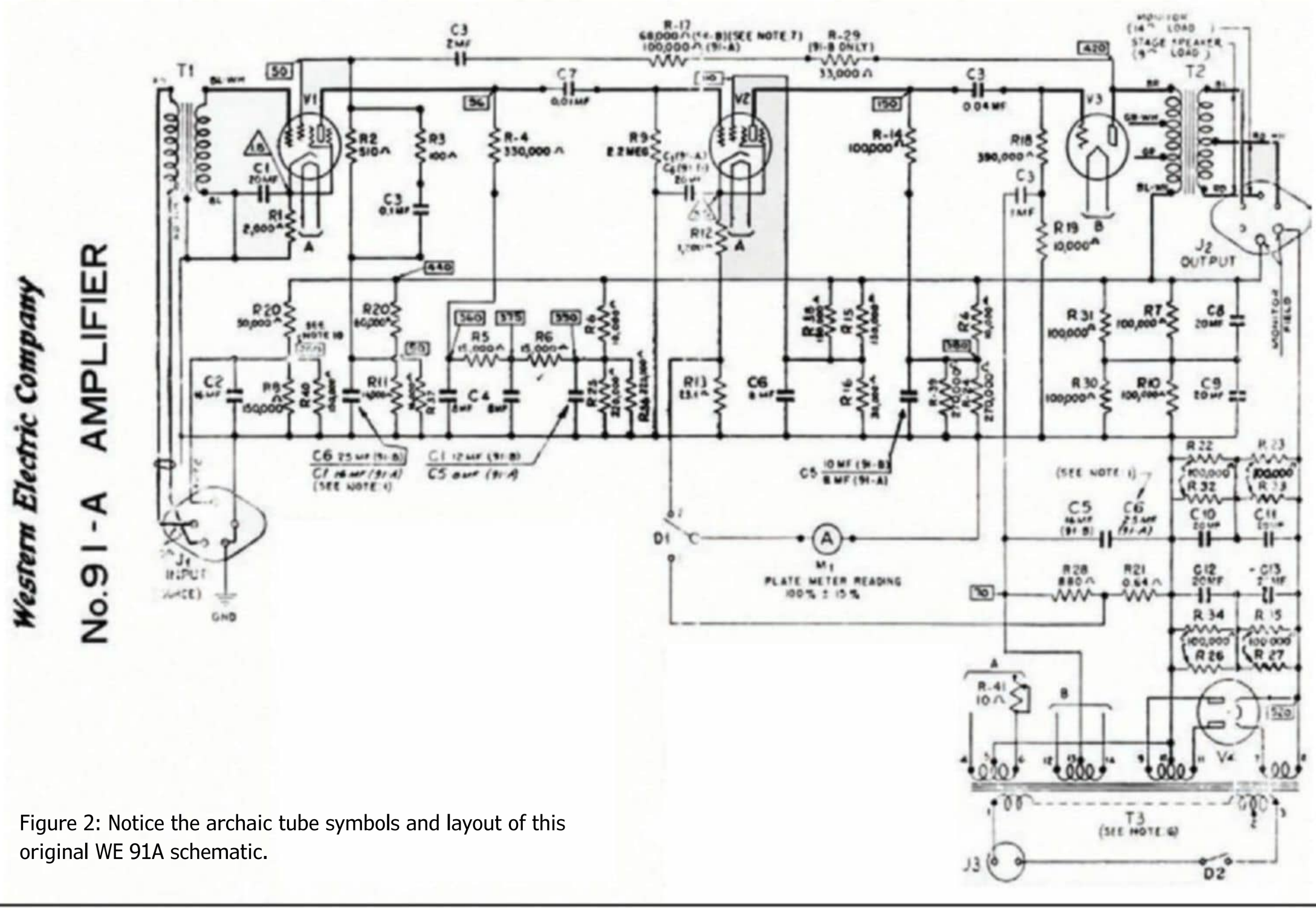

Figure 2: Notice the archaic tube symbols and layout of this original WE 91A schematic.

designers who use circuit modeling. So why would one want to do so?

First, modeling a circuit allows one to compare different circuit topologies without investing the time to build multiple iterations of a new project. Second, it provides a sort of "prototype prototype" that can provide a sanity check for a new design.

The downside is the aforementioned gremlin: human error. But truth be told, most of us have made errors in building physical circuits, and those can often be more costly than a modeling error.

A Regrettably True Story

In the September 2022 Hollow-State Electronics column, I compared the performance of a Western Electric (WE) Model 91A using 300 B triodes against a push-pull amp with no overall negative feedback. I based the comparison on an LTspice model of the WE 91A. Recently, reader Hans Weedon, an excellent engineer and careful reader, noted an error in my model, leading to erroneous results.

The error was that, as shown in Figure 1 of that Hollow-State Electronics article, I accidentally omitted the grid-leak resistor in the second stage. In order to get an approximation of the output amplitude and distortion I expected, I followed up by making another error: tweaking the component values in the model.

In my own defense, I will point out that I decided to tweak the values only after reviewing the model numerous times, each time not noticing the omission of the grid-leak resistor. This is certainly not an excuse, but it does point out how susceptible we humans are to blindness regarding our own mistakes.

In reviewing my LTspice model in preparation for the current article, I noticed several other mistakes in my previous model. The corrected model is shown in **Figure 1**, and it may be compared with the Western Electric schematic for the Model 91A amp shown in **Figure 2**. You will notice one of the factors that encouraged my errors: The original schematic used different symbols for the tubes than have been common for the last 70-plus years, leading differences in layout between this and a typical modern schematic, and to an increased opportunity to misread that schematic.

Figure 3 shows the 13W output waveform predicted by the new simulation. The slight clipping on the negative half-cycles is barely noticeable; except for this clipping, the waveform is nearly perfect. The FFT shown in **Figure 4** reveals a THD of 9%, primarily third harmonic. The combination of slight clipping and negative feedback result in a "forest" of higher harmonics, but the levels of harmonics above the sixth are 3dB or more below that of the second harmonic, with the level dropping about 40dB/decade as harmonic number rises.

I would like to reiterate my thanks to Hans Weedon for his attentive reading and for taking the time to bring my error to my attention.

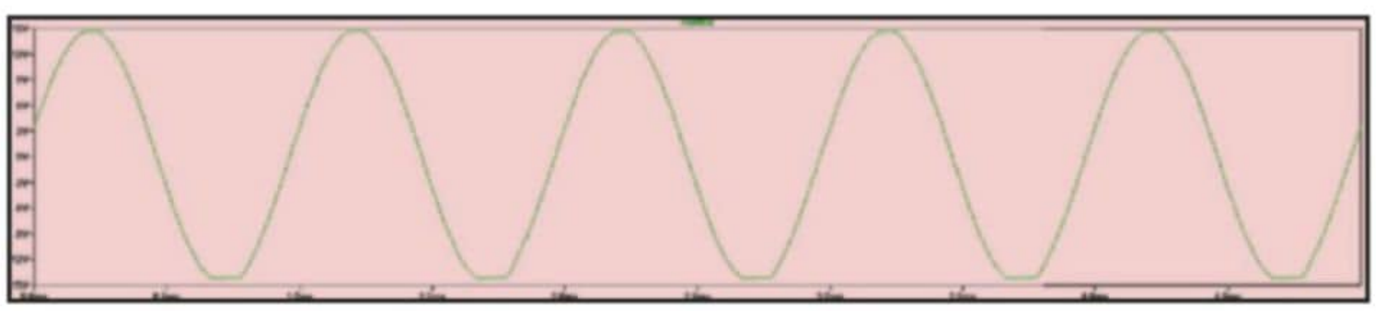

Figure 3: The output wave at 13W is an almost perfect sine wave.

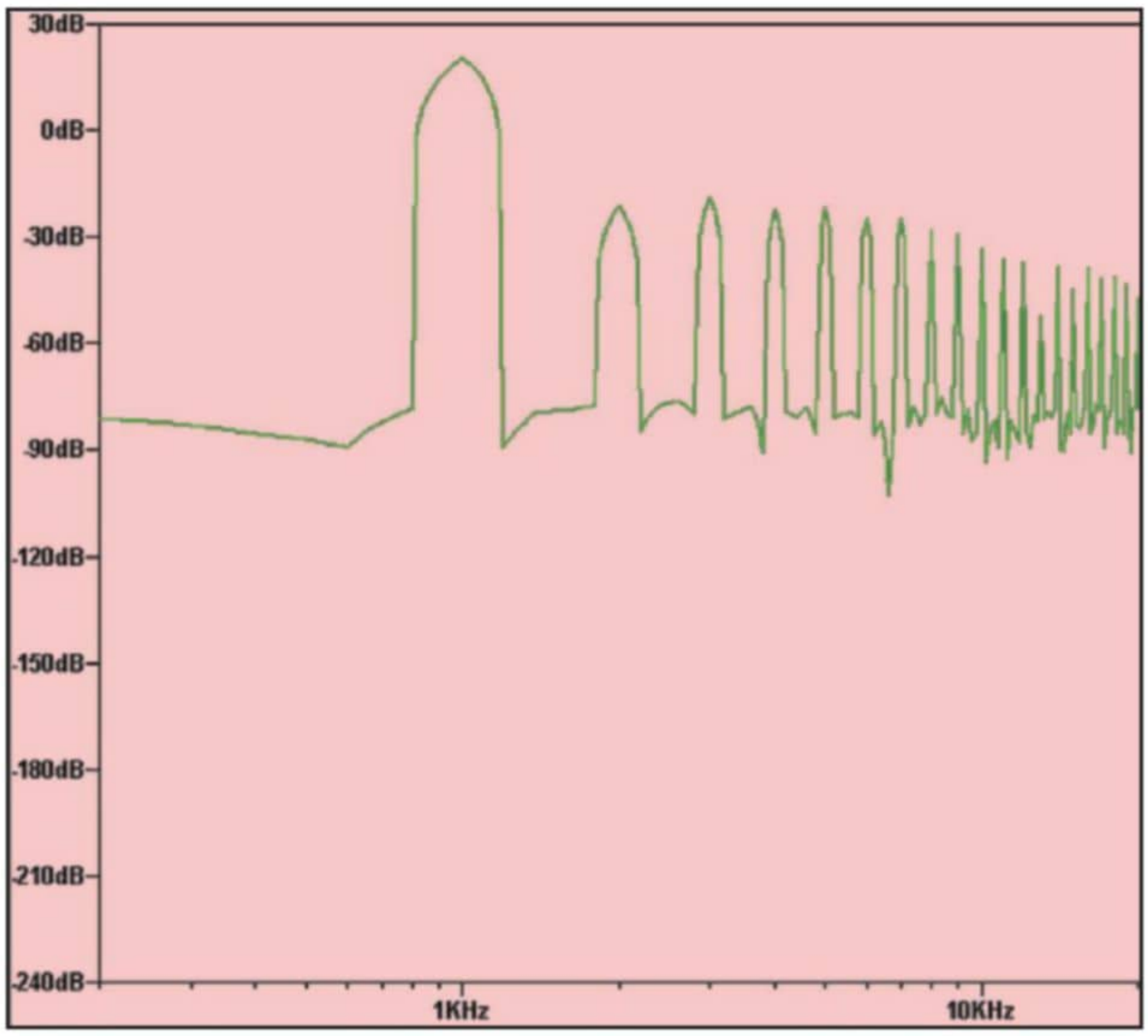

Figure 4: Third-harmonic distortion predominates.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.

Xpress Mail

RIAA Networks

I enjoyed John Ellis' article in *audioXpress* January 2022, "An Approach to Designing RIAA Networks." It has been more than 30 years since I used Newton-Raphson or differential equations to solve these issues (and that was in the financial arena).

Using an "ideal" operational amplifier and Laplace transform I developed the graph (**Figure 1**), which illustrates the errors of the Lipschitz equations compared with John's optimization. I think it better demonstrates the value of his work. Each trace is adjusted to 0dB error at 1kHz.

John's work eliminates the potential necessity of an additional low-pass filter in the compensation scheme.

—Jack Walton

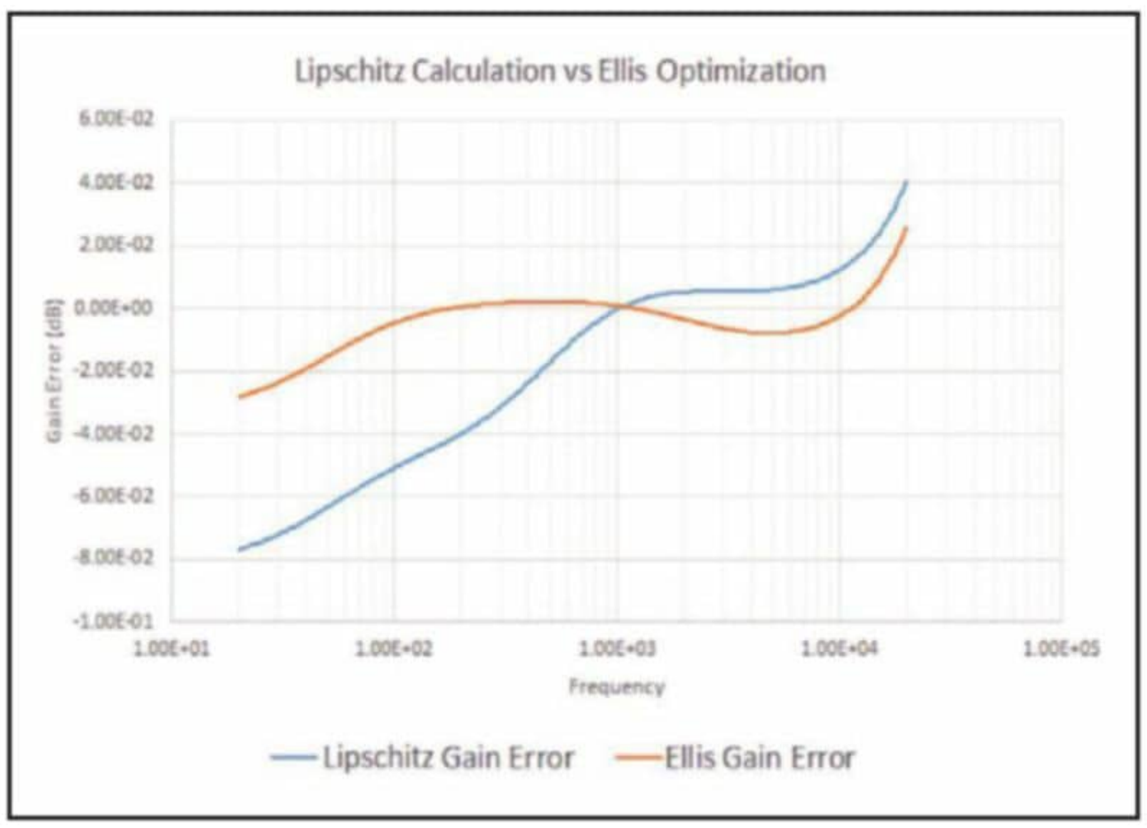

Figure 1: Lipschitz calculation vs. Ellis optimization

John Ellis Response:

I would like to thank Mr. Walton for his interest in my article on RIAA design and welcome his result in his evaluation. The overall response of a circuit with an RIAA feedback network depends on both the network and the amplifier around which it is applied. As I mentioned in the article, Newton's method is capable of very accurate results and with an ideal op-amp it was not a surprise that Mr. Walton found a good fit. In comparison to Lipschitz's results, I made a comparison to indicate equivalence. It surprised me more that Mr. Walton's result with Lipschitz's optimization was not closer.

One reason for writing the article was to show that the method can optimize the performance with less-than-ideal amplifiers, and to this extent depends on the accuracy of the transfer function used to describe the amplifier employed. It is fairly straightforward to include a transfer function in the numerical evaluation, but generally analytical expressions become more complicated.

Whether a low-pass filter is needed or not depends on the configuration. In the inverting mode, a low-pass filter is not required, but in the more usual non-inverting configuration, (which is usually chosen for lower noise) an amplifier will roll off to unity gain unless or until it reaches its own unity gain frequency. For greatest conformance to the RIAA curve a low-pass will usually be required on the output of these circuits, except, perhaps, if the frequency response of the preamp itself or other stages mitigates the need.

A second reason for writing the article was to inform those who may be familiar with Newton for single variable functions but less so for solving problems involving several variables, even though it is a well-known method to those working with numerical calculations. I should perhaps have made it clearer that the differentials are partial, so evaluating them numerically is relatively straightforward.

—John Ellis

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